Reason Studios

Reason 12

Rack Plugin

Operation Manual

Reason Studios

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Chapter 1
Introduction



Welcome!

This is the Operation Manual for Reason Rack Plugin, part of the Reason Version 12 music production software from Reason Studios. The information in this manual is also available as html files in the on-line Help system.

If you're using Reason mainly as plugin in another DAW host, this is the manual for you! If you're using Reason as a standalone music application in itself, you should check out the main Reason 12 Operation Manual.

Also, be sure to regularly check out www.reasonstudios.com for the latest news!

! **The information in this document is subject to change without notice and does not represent a commitment on the part of Reason Studios.**

Introducing Reason Rack Plugin

Reason Rack Plugin is an instrument and effect plugin in VST3, AUv2 (Mac) and AAX formats. Reason Rack Plugin is installed when you install the standalone Reason program.

Like the standalone Reason 12 application, Reason Rack Plugin works on two platforms: Windows 10 (64-bit) and macOS 10.13 (High Sierra) (64-bit) or later. All functions are the same. If your DAW host supports cross-platform functionality (saving a project on one platform and opening it on another), Reason Rack Plugin will open and work the same on both platforms (provided of course that Reason is installed on both computers).

The screenshots in this manual were taken from both platform versions of Reason Rack Plugin. Since the layout is more or less identical in these versions, there shouldn't be any problem following the instructions.

Conventions in the manual

This manual describes both the Windows and macOS versions of Reason Rack Plugin; wherever the versions differ this is clearly stated in the text.

Text conventions

The text conventions are pretty straightforward. The examples below describe when certain text styles are used:

→ **This style instructs the user to perform the task(s) described in the sentence.**

! **This text style means IMPORTANT INFORMATION. Read carefully to avoid problems!**

► **This text style is used for tips and additional info.**

Key command conventions

In the manual, computer keyboard commands are indicated with brackets. For example:

→ **Hold down [Shift] and press [C].**

However, some modifier keys are different on Windows and Mac computers. Whenever this is the case, the manual separates the commands with “(Win)” and “(Mac)” indications as in the following example:

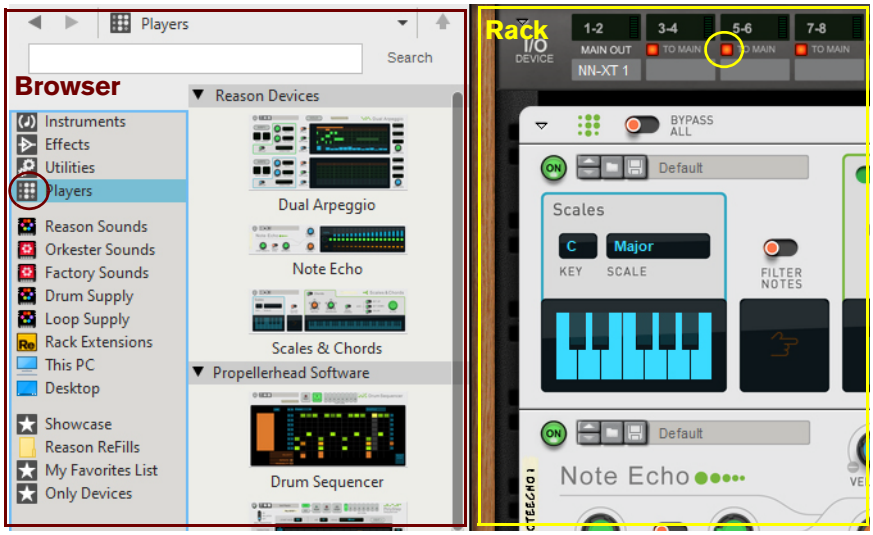
→ **Hold down [Ctrl](Win) or [Cmd](Mac) and press [C] to copy.**

References to context menus

Whenever the manual instructs you to select an item from the “context menu”, it means that you should right-click (or [Ctrl]-click if you’re using a Mac with single-button mouse) on the specific area, section or device, and then select the item from the pop-up menu that appears - the context menu. The item list in context menus varies depending on where in the application you click.



Frames and circles (call-outs)



In pictures throughout this manual there might be circles and/or rectangles highlighting certain areas or objects. These are indicated by filled lines according to the examples in the picture above. Sometimes these highlighting frames/circles might also be accompanied by descriptive texts. The different colors of the frames and texts are only to enhance the contrast to the background pictures.

Dashed arrows



A dashed arrow in a picture indicates the directions in which the pointer (or other tool) should be dragged to perform the desired operation. The example in the picture above shows in which directions (up and down) to drag the pointer to change the knob's setting.

About the Authorization system

! **Note that the information below is for Reason 12.6 and later. The information applies both to Reason+ subscriptions and to perpetual licenses, unless noted.**

Reason Rack Plugin is authorized in the same way as the standalone Reason application, and uses the same license or Reason+ account.

- **The core of the authorization system is your Reason Studios account (and license number, if you are using a perpetual version of Reason 12).**

Registering Reason (perpetual version only)

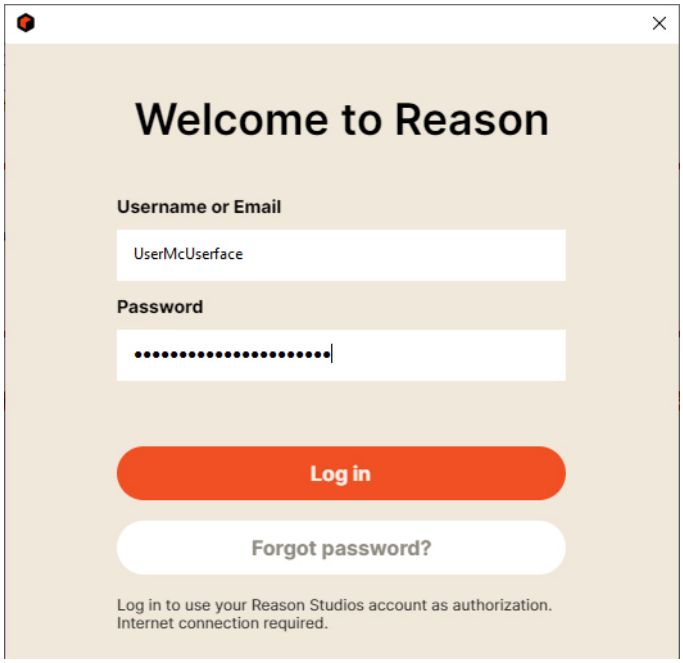
To be able to run your perpetual (licensed) version of Reason Rack Plugin, Reason must be registered to your account on the Reason Studios web site.

→ **If you purchased a Reason license in a music store, follow the instructions you received from the shop.**

! **If you purchased Reason directly from the Reason Studios Shop web site, the Reason license has already been automatically registered to your user account and you can use Reason by running it on a computer connected to the Internet.**

Authorizing Reason Rack Plugin

! **Note that Internet connection is initially required for the procedures described below.**



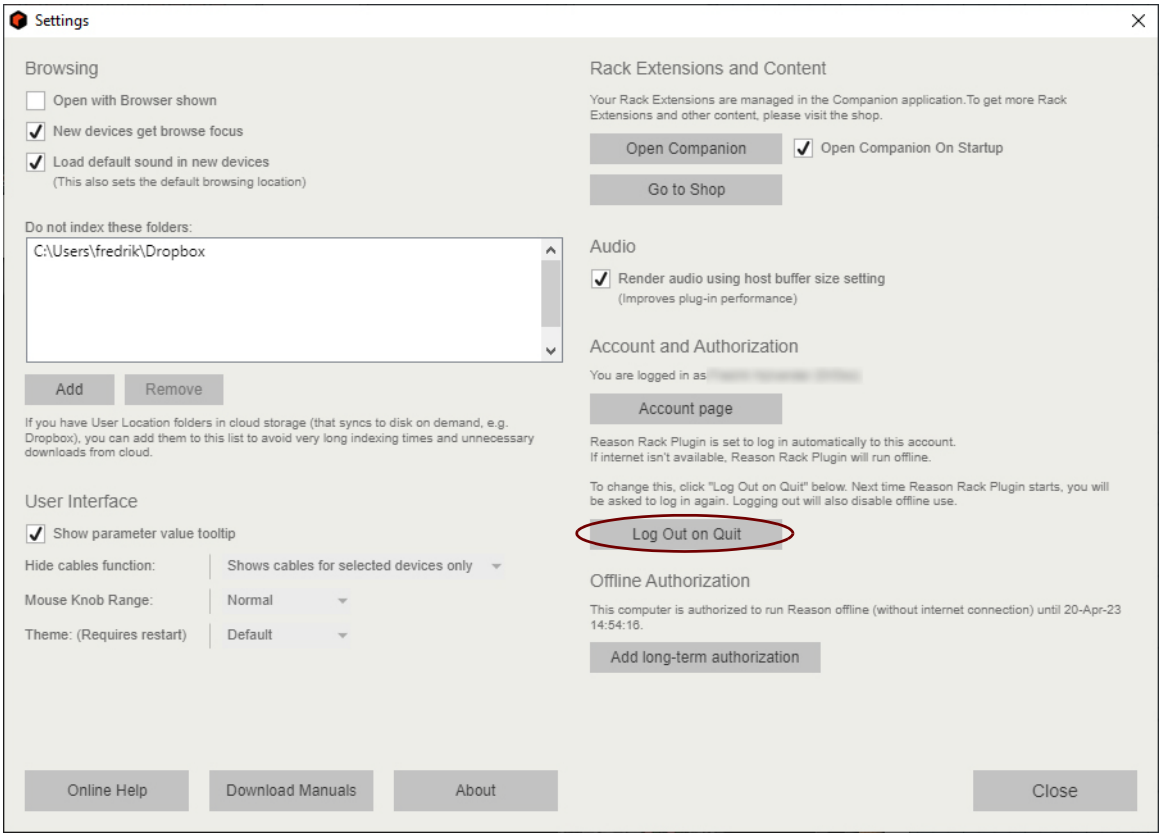
1. **Launch your DAW and create a Reason Rack Plugin instance.**
2. **Enter you Username/e-mail address and Password for your Reason Studios account and click the Log in button.**
 Reason Rack Plugin launches in Authorized Mode.

- **The first time you log in to the "Welcome to Reason" login dialog, the program will automatically store your user credentials.**
 The next time you create a Reason Rack Plugin instance, it will start without showing the login dialog. This also automatically enables off-line use, allowing Reason Rack Plugin to run even if you have no Internet connection. Reason Rack Plugin will then run off-line for a month. This period is continuously renewed when you use Reason Rack Plugin (perpetual or Annual Reason+ subscription) with an Internet connection. You do not have to do anything manually, except connecting to the Internet once a month.

Logging out from Reason Rack Plugin

Logging out means removing the stored login credentials - do this if you're not going to use Reason Rack Plugin on that computer, or if you need to log in to another Reason Studios user account.

➔ **In the Settings dialog in Reason Rack Plugin, click the "Log Out on Quit" button:**



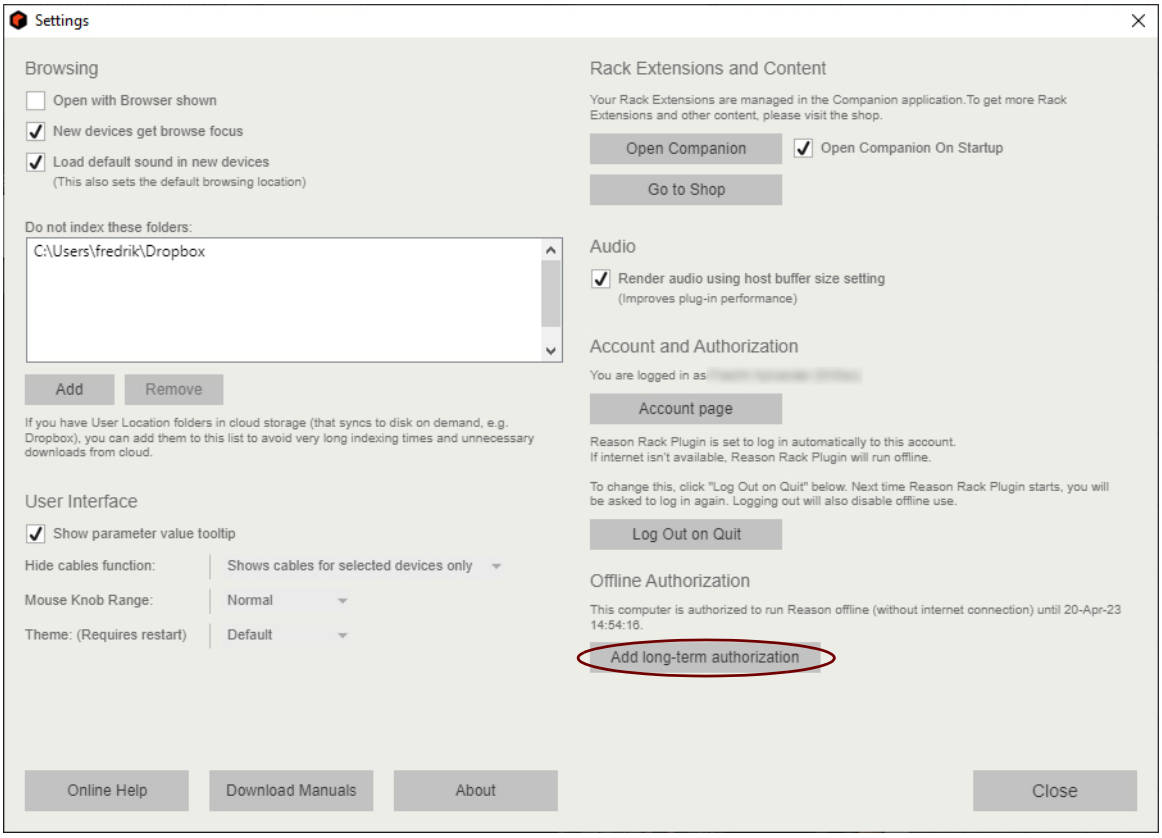
The next time you run Reason, you will need to enter your Username/e-mail and password in the "Welcome to Reason" dialog again. Internet connection is then required again.

Authorizing a computer for long-term off-line use

In addition, if you have an Annual (yearly) Reason+ subscription or own a perpetual (licensed) version of Reason 12, you can authorize your computer for long-term off-line use. This can be useful if you are traveling or working in a studio without Internet access, etc. The long-term authorization allows for staying off-line on the authorized computer for one (1) year. If you are a Reason+ subscriber you are limited by the subscription end date, though. In most situations, authorizing a computer for long-term use is not necessary, though.

Authorizing Reason Rack Plugin for long-term off-line use

1. **Open the Settings dialog.**
2. **Click the "Add long-term authorization" button:**



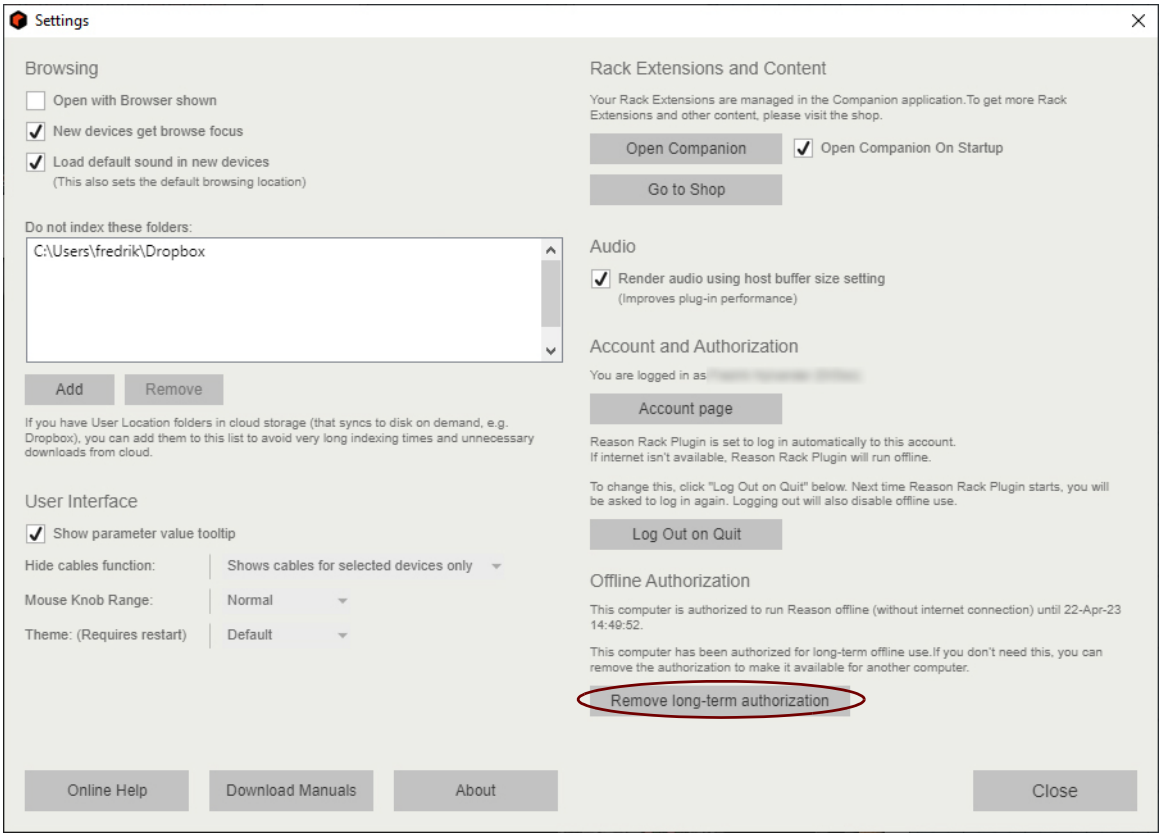
- **The authorization expiry date for your computer is shown in the Offline Authorization section.**
- **You can authorize up to three (3) computers for long-term off-line use.**

Removing a long-term authorization for a computer

If you've already long-term authorized three computers and would like to authorize another computer, you will need to remove the authorization for one of the computers first:

1. **Launch your DAW and add a Reason Rack Plugin instance on one of the computers that already has a long-term authorization.**
2. **Open the Settings dialog.**

3. **Click the “Remove long-term authorization” button:**



- **If you can no longer access any of your authorized computers, you need to contact support to have them remove the authorization(s). This can be done here.**

What happens if an authorization expires?

- **If a Reason Rack Plugin authorization should expire during a session, you can continue to work and save your song as usual.**
- **The next time you open Reason Rack Plugin you will have to log in in the “Welcome to Reason” dialog again.**

About Rack Extensions

Rack Extensions are additional devices that can be purchased or trialed from the Reason Studios web shop. Rack Extensions can be instruments, effects or utility devices, such as mixers and CV processors. Rack Extension devices are developed by Reason Studios as well as by 3rd party companies.

Once installed, Rack Extensions will be available both in standalone Reason and in Reason Rack Plugin. In the program or plugin, they behave just like built-in devices.

- **To browse, trial or purchase Rack Extensions, visit reasonstudios.com/shop**
- **To download and install Rack Extensions that you own, visit your user account page.**
- **To manage your installed Rack Extensions, use the Reason Companion app, which can be downloaded here.**

You can open all these directly from the Rack Extensions section in the Settings dialog, see "Rack Extensions and Content".

The "Update Rack Extension Licenses" alert

Your Rack Extensions are authorized in the same way as Reason Rack Plugin. However, if you are opening the plugin when connected to the Internet, you may get an alert asking you to "Update Rack Extension Licenses". This means the Rack Extension license on your user account has changed, and no longer matches the license components on the computer (for example if you have added new Rack Extensions to your account, or if Trial licenses have timed out). Choose whether to update the license components on your computer or to run this session without Rack Extensions.

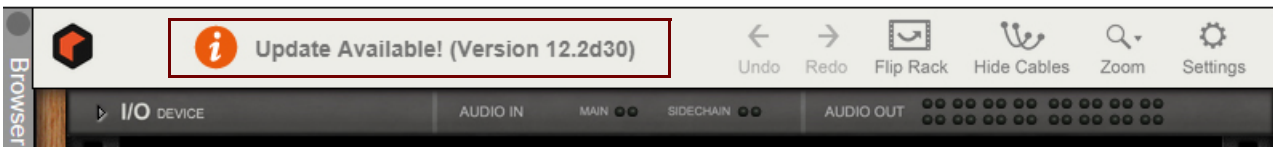
Getting all your content

The main sound banks are installed when you install Reason, as are a great many instrument and effect devices. However, there is also additional, optional content that you should check out:

- **Reason and Reason Suite include the Drum Supply and Loop Supply ReFills and four additional instrument devices (Radical Piano, Klang Tuned Percussion, Pangea World Instruments and Humana Vocal Ensemble).** To download these, you need to launch the stand-alone version of Reason and use the Manage Content function on the Window menu. After installation, these ReFills and devices will be available the next time you open Reason Rack Plugin.

About automatic update checks

When you launch Reason Rack Plugin it automatically checks for new updates on the Reason Studios web site (provided that your computer has Internet connection). If a new update is found, an alert will be shown on the Global Panel at the top of the Reason Rack:



- **Clicking the alert will launch your web browser and download an installer for the new version.** Once download is complete, quit the DAW and run the installer to update Reason Rack Plugin.

Alternatively, you can download and install the update directly from within the standalone version of Reason. This will also update the Reason Rack Plugin.

Chapter 2
Overview

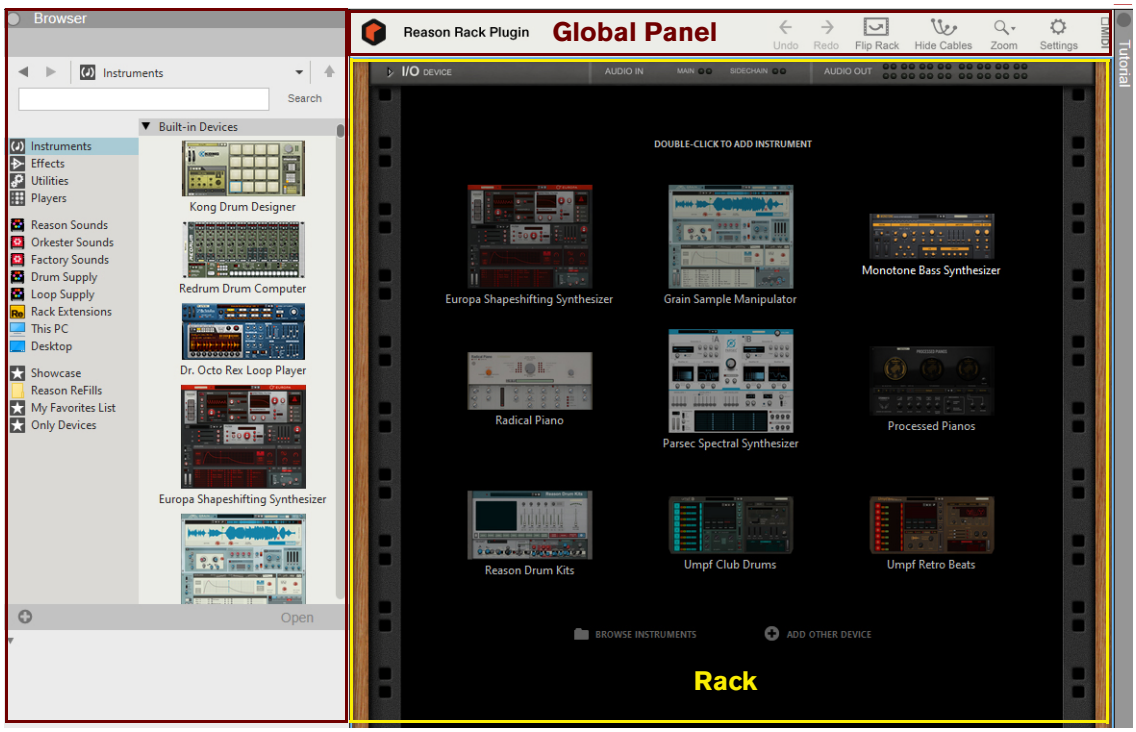


Adding Reason Rack Plugin in your project

Reason Rack Plugin comes in two flavors: Reason Rack Plugin (for use as an instrument) and Reason Rack Plugin Effect (for use as an audio effect, processing the sound from other instruments or audio tracks).

→ **Add Reason Rack Plugin instances to your project like you would with other VST plugins.**
You can add as many instances of Reason Rack Plugin as your computer can handle.

The Reason Rack Plugin window



The Reason Rack Plugin has three main areas:

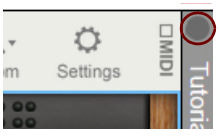
- **The Browser**
 The Browser is where you create devices, browse patches and load samples. This area is folded by default, but you can unfold it by clicking the circle button in its top left corner:

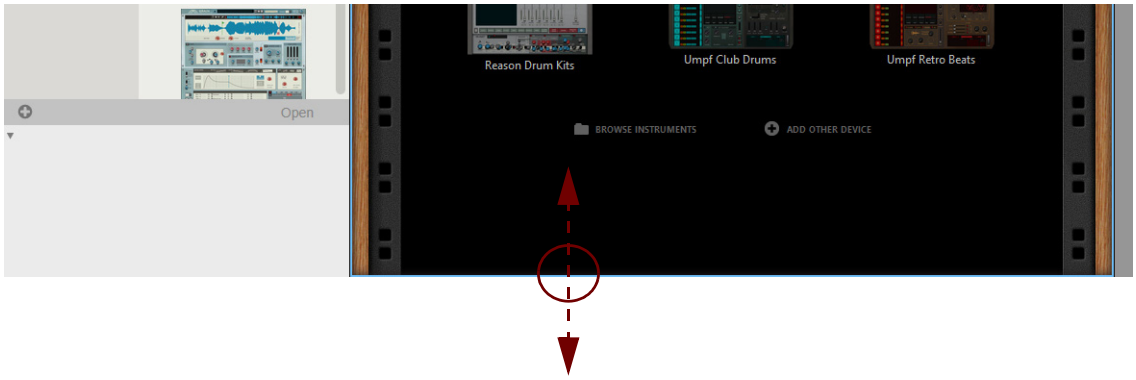
 It will also automatically appear when you click the Browse button on a device. See the "Sounds, Patches and the Browser" chapter for more about browsing patches and samples.

- **The Rack**
 The Rack is where you build your sound, by placing instrument and/or effect devices.

- **The Tutorial area**
 This contains tutorials showing you how to use Reason Rack Plugin. You can fold and unfold this area by clicking the circle button in its top corner:

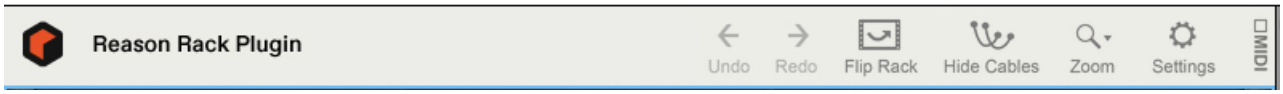
 

The Reason Rack Plugin window can be resized vertically by dragging the lower window edge, which is quite useful if you add many devices to your rack (and have a large monitor).



Scrolling in the rack is done by using scroll wheel, Page Up/Down buttons on your computer keyboard or by clicking and dragging the side panels up or down.

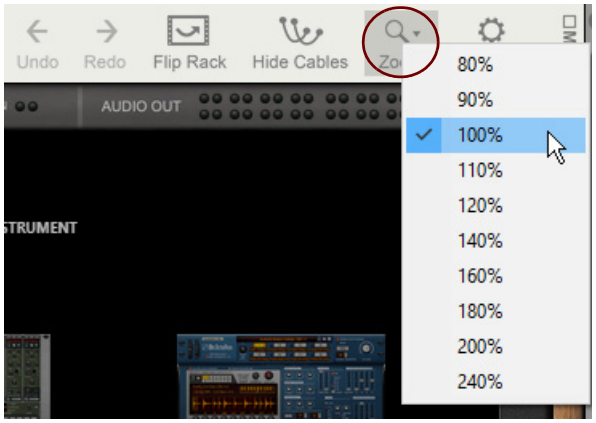
Above the rack you'll find the Global Panel, which holds some important functions such as Undo/Redo, Flip Rack and a button for opening the Settings dialog. To the right is also a MIDI In indicator:



Zoom

It's possible to choose a suitable zoom level for the Reason Rack Plugin. This can be useful if you are using a very high-resolution screen and want to make Reason Rack Plugin larger. Since all built-in rack devices and most Rack Extensions are high-resolution, you will not lose any image quality when you enlarge Reason Rack Plugin.

→ **Click the Zoom button and select the desired Zoom factor:**



! **Note that the first time you select a new Zoom level, all devices in the song will recalculate their high-resolution panels in the background, which might take a little while.**

About different Themes

In Reason Rack Plugin you can choose from a couple of different visual themes, i.e. how the user interface should be visually presented. The selected theme affects the Browser, Global Panel, Settings dialog and Tutorial area.

To select another Theme, click the Settings button and select an option from the Theme menu in the dialog that appears. For the changes to take effect, you need to quit and relaunch your DAW host.

Editing parameters

Since devices in Reason Rack Plugin are largely laid out like "real" hardware synths and effects, almost all controls are designed like their real world counterparts - mixer faders, effect unit knobs, buttons, etc. How to adjust these controls is described in the following paragraphs.

Knobs

- **To “turn” a knob, point at it, hold down the mouse button and drag up or down (as if the knob was a vertical slider).**
 Dragging upwards turns the knob clockwise and vice versa.



- **If you press [Shift] and drag, the knob will turn slower, allowing for higher precision.**
 You can also adjust the knob precision with the “Mouse Knob Range” setting in the Settings dialog.
- **To reset a knob to its default value (usually zero, center pan or similar), press [Ctrl](Win) or [Cmd](Mac) and click on the knob.**

Faders and sliders

- **To move a fader or slider, click on the handle and drag in the fader/slider direction.**



- **You can also click anywhere on the fader/slider to instantly move the handle to that position.**
- **If you press [Shift] and drag, the fader/slider will move more slowly, allowing for higher precision.**
- **To reset a fader/slider to its default value (usually zero, 100, center pan or similar), press [Ctrl](Win) or [Cmd](Mac) and click on the fader/slider handle.**

Buttons

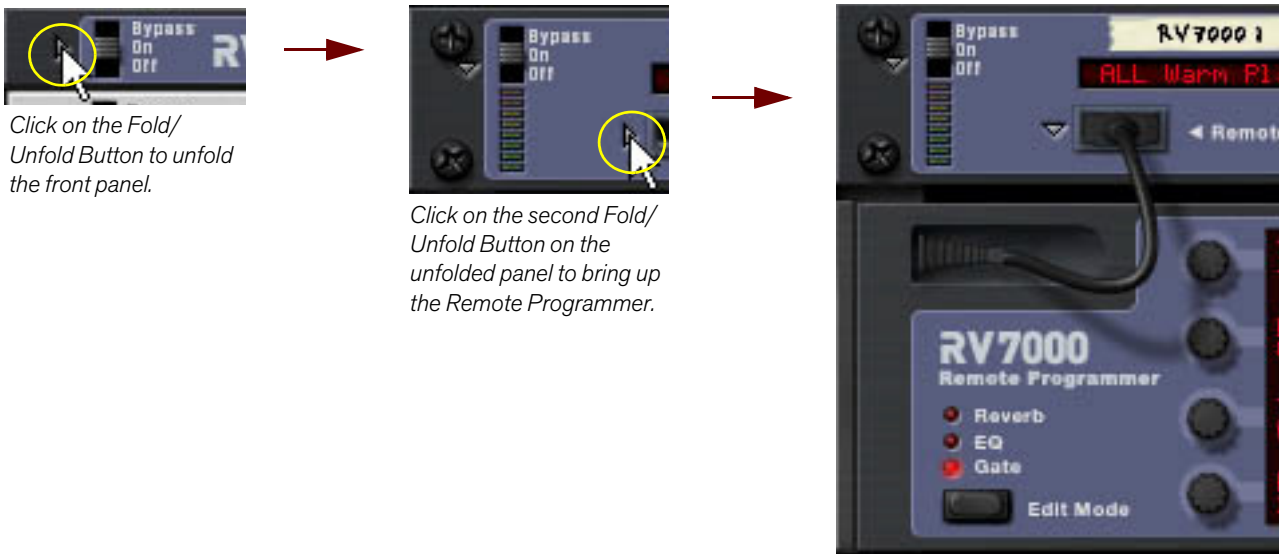


Many functions and modes are controlled by clicking buttons. Many of the buttons in Reason have a “built-in” LED, or the button itself lights up, indicating whether the button is on or not.

Fold/Unfold buttons

Fold/Unfold buttons are distinguished by a small triangle at the top to the left on a device. Clicking on a Fold/Unfold button will unfold the device panel so that more controls are visible and can be accessed for editing on the screen.

On some devices, such as the RV7000 Advanced Reverb, there are more than one Fold/Unfold button. Clicking on the second Fold/Unfold button on the unfolded front panel will open up the Remote Programmer panel from which more parameters can be accessed:



Click on the Fold/ Unfold Button to unfold the front panel.

Click on the second Fold/ Unfold Button on the unfolded panel to bring up the Remote Programmer.

The Fold/Unfold buttons on an RV7000 Advanced Reverb device.

- **Pressing [Alt](Win) or [Option](Mac) and clicking a Fold button will fold or unfold all devices in the rack.**

Multi Mode selectors

Some parameters allow you to select one of several modes. There are two different graphical representations of this in Reason.

The multi mode selector type below consists of a button with the different modes listed above it:



→ **Click the button to step through the modes or click directly on one of the modes printed on the panel, or click on the corresponding LED, to select mode.**
The currently selected mode is indicated by a lit LED.

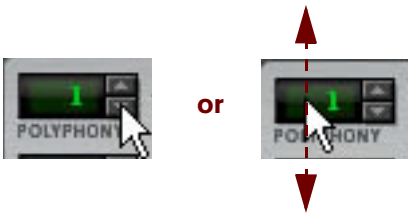
The multi mode selector type below is a switch with more than two positions:



→ **To change mode, click and drag the switch, or click directly at the desired switch position (just as when adjusting a slider).**

Numerical controls

In Reason devices, numerical values are often displayed in numerical displays with “spin controls” (up/down arrow buttons) on the side. Some parameter values, such as oscillator and LFO waveforms, are displayed graphically in the displays. There are two ways of changing values in these types of controls:



- **By using the up and down buttons on the spin controls.**
 To adjust a value in single steps, click on its up or down arrow button. To scroll a value continuously, click on an arrow button and keep the mouse button depressed.
- **By clicking and holding the mouse button depressed in the actual display and then dragging the mouse up or down.**
 This allows you to make coarse adjustments very quickly.

Menus and alpha-numeric controls

In Reason, alpha-numeric values and/or device presets are displayed in alpha-numeric readouts with “spin controls” (up/down arrow buttons) on the side. There are two ways to change alpha-numeric/preset values:



- **By using the up and down buttons on the spin controls.**
 To scroll continuously, click on an arrow button and keep the mouse button depressed.
- **By clicking and holding the mouse button depressed in the actual alpha-numeric display and selecting from the list that appears.**
 This allows you to make coarse adjustments very quickly or to immediately change to a preset anywhere in the list.

- **This type of control is used to select, e.g., patch and reverb algorithms and some oscillator waveforms.**

Tool Tips

If you hover with the mouse over a control on a device panel and wait a moment, a tool tip appears. The tool tip shows the name of the parameter associated with that control and its current value. This helps you fine-tune settings, set several parameters to the same value, etc.

- **You can turn off the Tool Tips function by deactivating the option “Show parameter value tool tip” in the Settings dialog (see “Show parameter value tool tip”).**

Context menus

Context menus are “tailored” to contain only menu items that are relevant to the current circumstances. Using the various context menus allows you to work more quickly and more efficiently with Reason Rack Plugin.

→ **To bring up a context menu, right-click on the desired object, section or area in Reason Rack Plugin.**
If you're using a Mac with a single-button mouse, press [Ctrl] and click.

The contents of the context menus depend on where you click. These are the primary types of context menus you will encounter in Reason:

Parameter context menus

Depending on the host DAW implementation there may or may not be parameter context menus. If available, the host DAW defines what items are available.

Device context menus

If you context-click anywhere on a device in the Rack (but not on a parameter or display), the context menu will contain the following items:

- **Undo and Redo functions.**
- **Cut, Copy, Paste, Delete and Duplicate Device items, allowing you to rearrange and manage the devices in the rack.**
- **Submenus for creating new devices (Instruments, Effects, Utilities or Players).**
- **Combine and Uncombine are used when you want to use the selected device in, or exclude it from, a Combinator setup.**
- **Auto-routing and Disconnect functions.**
 This allows you to automatically route (connect) or disconnect a selected device in a logical way.
- **Reset Device resets all parameters to their default values and removes any loaded samples.**
- **If the device supports patches, the menu contains functions for copying, pasting and browsing patches.**
- **At the bottom of the menu are Flip Rack and Hide Cables items.**
 These work the same as the buttons on the Global Panel at the top of the rack.
- **Additional device-specific items.**

There may also be additional device-specific functions, for managing samples, handling patterns and more.

Empty rack context menu

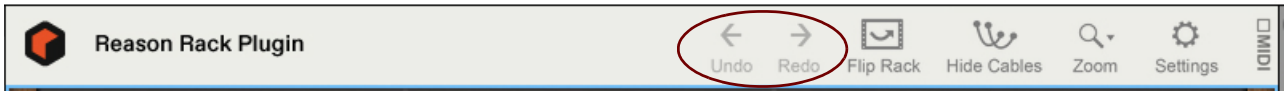
If you click in an empty area of the rack, the context menu will contain the following items:

- **Undo/Redo.**
- **A Paste item, allowing you to paste any copied or cut devices.**
- **Functions for creating new devices.**
- **Flip Rack and Hide Cables.**

Undo and Redo

While virtually all DAW hosts have Undo and Redo functions, many don't allow you to undo changes done within plugins. This means that you might create a plugin instrument, change some parameters in the plugin and select undo - only to have the program remove the plugin you created in the first step. The parameter changes aren't part of the DAW host's undo history.

To avoid this, Reason Rack Plugin has its own Undo and Redo functions. These are available as buttons on the top Global Panel, and as functions on the context menus:



! Each instance of Reason Rack Plugin has its own Undo history.

Chapter 3
Audio and MIDI Basics

General audio and MIDI handling

Reason Rack Plugin doesn't communicate directly with your audio or MIDI hardware. Instead, this is handled by your DAW host, which in turn passes on MIDI or audio to Reason Rack Plugin and gets audio back in return.

A Reason Rack Plugin instance can:

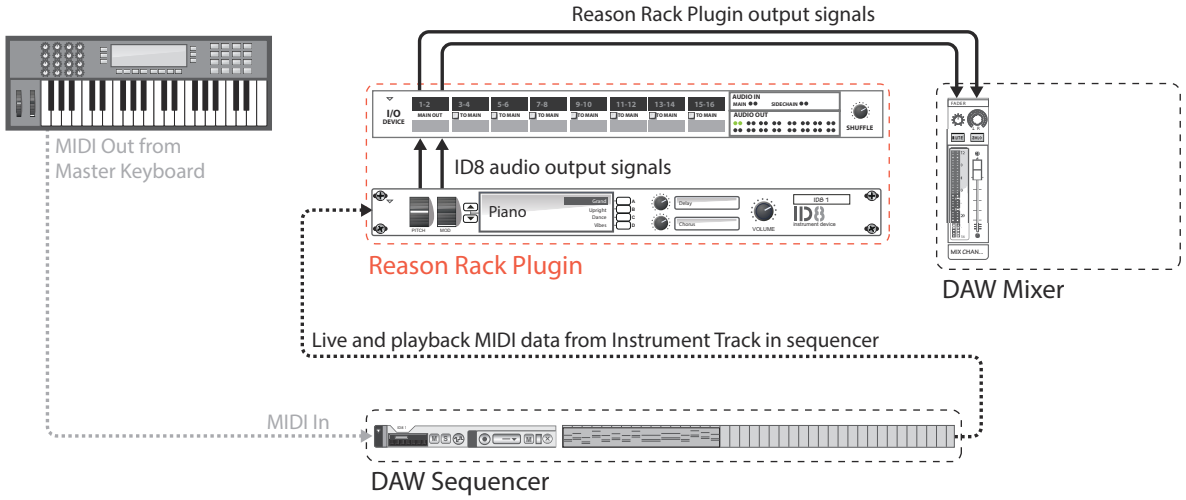
- **Receive MIDI notes and other messages from the DAW host. It does not care about MIDI channels.**
- **Receive up to four audio channels (two stereo input pairs).**
 Typically, Reason Rack Plugin receives audio when used as an audio effect, but it's also possible to send audio to an instrument (if your DAW permits this), for sidechaining and other effects.
- **Send out up to 32 audio channels (16 stereo pairs) to the DAW host.**
 In most cases, the output will be a single stereo signal, but you may for example want to route different drum sounds to different outputs for processing on separate channels in your DAW host's mixer.
- **Reason Rack Plugin can also send MIDI to other tracks in your DAW - if you use the MIDI Out Device, see "MIDI Out Device".**

Typical input/output configurations

Below are some examples of typical MIDI and audio configurations in Reason Rack Plugin:

Stereo instrument device

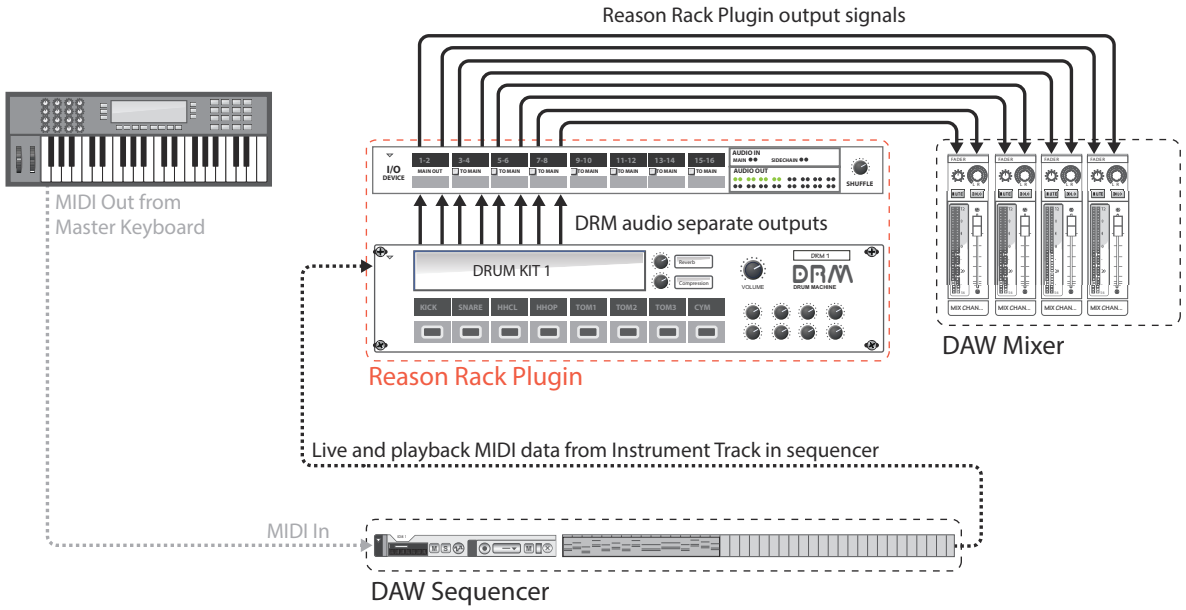
This setup involves MIDI note input sent to an instrument device, and stereo audio output sent from the instrument device:



Stereo instrument example.

Multi-channel instrument device

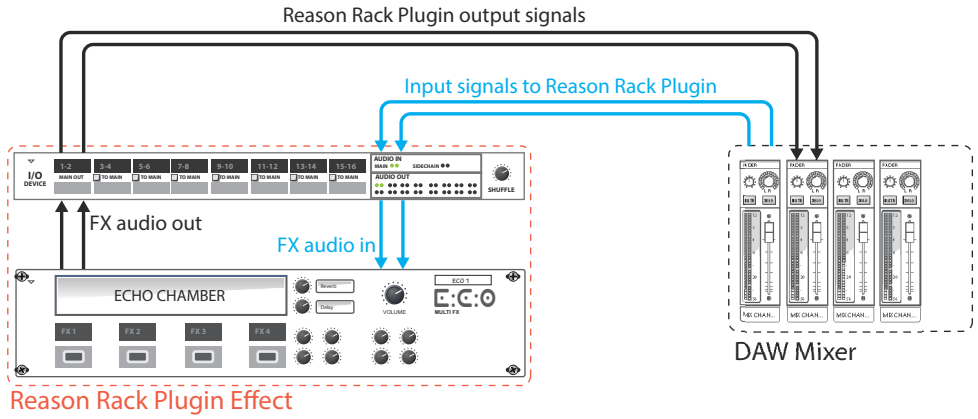
This setup involves MIDI note input sent to an instrument device, and audio sent out from multiple audio outputs of the instrument device. A typical scenario would be a drum machine device with multiple drum channels:



Multi-channel instrument example.

Stereo audio effect device

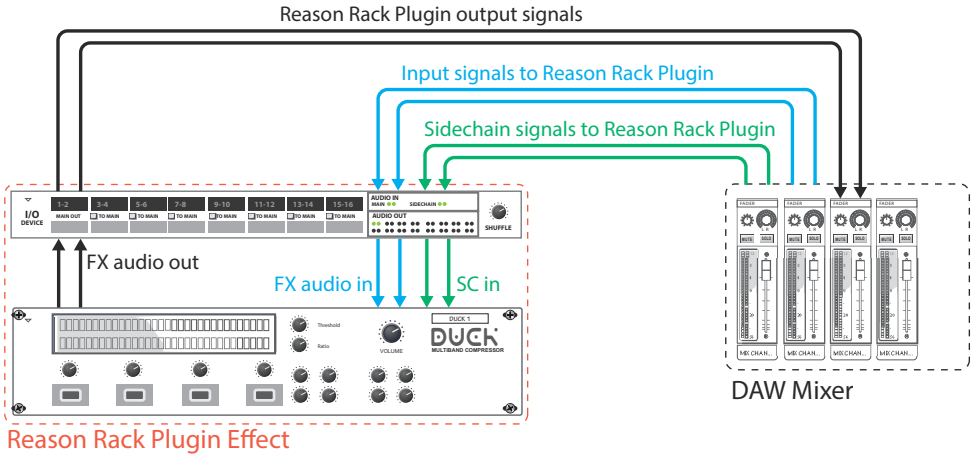
This setup involves a stereo audio input signal to an effect device, and stereo audio output from the effect device:



Stereo effect example.

Stereo audio effect device with sidechain

This setup involves a stereo audio input signal to an effect device - plus a sidechain audio input signal to the effect device - and stereo audio output from the effect device. A typical scenario would be a stereo compressor device with sidechain inputs.



Stereo effect with sidechain example.

The I/O device



The I/O device is always located at the top of the rack.

At the top of the rack is the i/o device (for "input/output"). This handles the audio communication between the devices in the rack and the DAW host.

The input jacks deliver audio from the DAW host to devices in the rack. This is most often the case when Reason Rack Plugin is used as an effect, and typically only the main (1-2) input jacks are used.

The output jacks deliver audio from devices in the rack to the DAW host. These are used both when Reason Rack Plugin is an instrument and an effect.

On the front panel you find audio input and output indicators, lighting up whenever audio signals are received from the DAW host (input) or sent back to the DAW host (output).

The first eight output pairs have more detailed meters, name labels matching the device names and buttons called (sum) “To Main" (outputs 3-16 only). When sum To Main is on for an output pair, its signal is directed to the Main Out (1-2) instead, and summed with any other signals there.

- **If you want to layer several instrument devices on a single stereo channel in your DAW, leave "To Main" on.**



The three instruments (connected to outputs 1-2, 3-4 and 5-6) will all be sent out on Main Out 1-2 to the DAW.

- **If you want different devices in your rack to be routed to different audio channels in your DAW's mixer, turn off "To Main" for these outputs.**

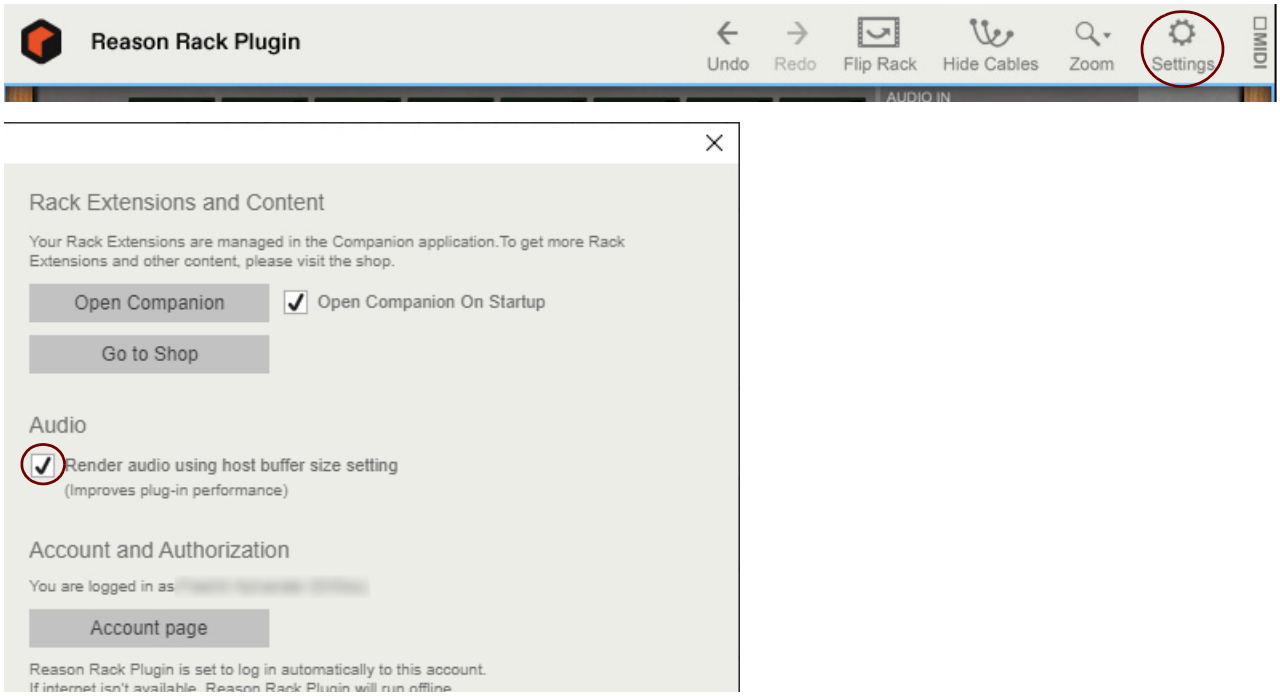


The two instruments connected to outputs 1-2 and 3-4 will be sent out on Main Out 1-2 to the DAW. The instrument connected to outputs 5-6 will be sent to the separate Out 5-6 outputs.

Audio settings

Since Reason Rack Plugin doesn't communicate directly with the audio hardware, audio settings like sample rate are all made in the DAW host. There is however one audio setting in the Settings dialog:

Render audio using host buffer size setting



When this is activated (default) all audio rendering will be done in batches corresponding to the buffer size selected in the DAW host's audio settings. Selecting a higher buffer size there will improve the performance of Reason Rack Plugin. However, if your rack contains feedback routings, these will be delayed with higher buffer sizes.

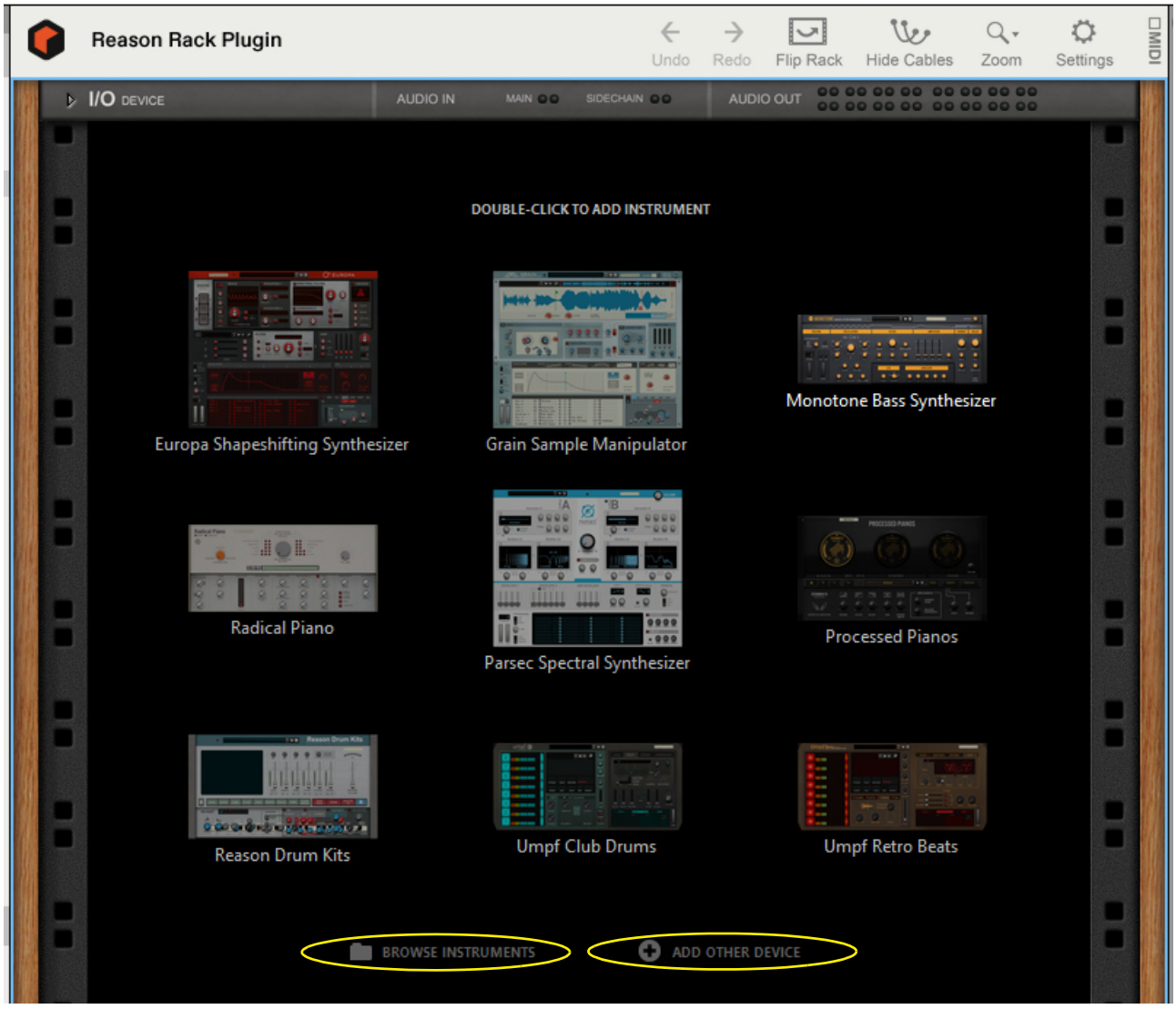
Turning this off will cause the plugin to render audio in batches of 64 samples (like in older versions of Reason). Use this only if you want to minimize delay in feedback routings in Reason Rack Plugin.

About Plugin Delay Compensation

There is no delay compensation made in the internal device routings of the Reason Rack Plugin instance itself. However, the summed delay in the signal chain to the Main Outputs 1-2 is reported to the host DAW, to allow for delay compensation against other tracks or channels.

Chapter 4
Using Reason Rack Plugin as an Instrument

Creating an instrument



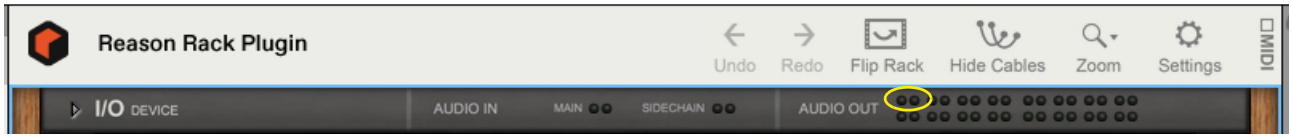
When you add Reason Rack Plugin as an instrument the plugin window opens. When the rack is empty, an overlay is shown with icons of the most popular instrument devices. Either:

- **Double click an instrument icon to add that instrument,**
- **click "Browse Instruments" to open the Browser with the Instruments palette shown,**
- **or click "Add other device" to add another device from the context menu that appears.**

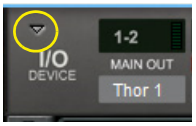
When you have added a device, the popular devices overlay goes away - it is only shown when the rack is empty. Note that the list of popular devices will change over time to include the instrument devices you most often add!

When an instrument device is added, it will automatically receive MIDI input from the track in the DAW host - you should be able to play it from your MIDI keyboard right away.

The output of the added instrument device is auto-routed to the first available output jack on the i/o device at the top of the rack.



1. **Click the arrow to the left to unfold the I/O device.**



2. **Click the Flip Rack button on the Global Panel to see the jacks on the back:**



A Thor instrument automatically connected to Main Out (1-2) of the I/O device.

Browsing for patches

Most instrument devices come with patches for quickly changing sounds. There are two main ways to select a patch for a device:

- **Use the patch name display on the device panel.**
 Either click and select a patch from the menu or use the previous/next patch buttons next to the display:



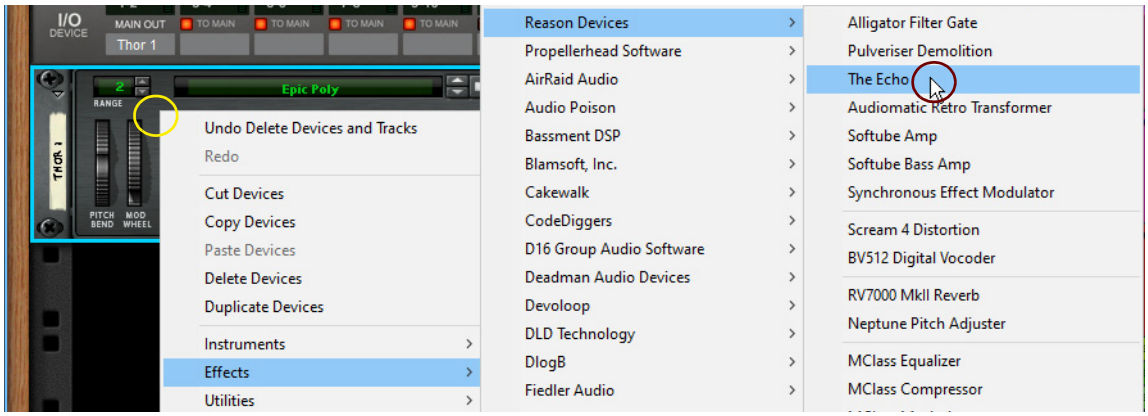
- **Click the Browse Patches button on the device (or select Browse Patches from the context menu).**



This opens the Browser, where you can search and navigate the sound banks and folders on your hard drive. Read more in the "Sounds, Patches and the Browser" chapter.

Adding effects

After having created an instrument and selected a sound for it, you might want to add one or several effects. You can either do this from the Browser or by context-clicking the instrument device and selecting an effect device from the Effects sub-menu:



The effect device is automatically connected, so that the audio from the instrument is routed through the effect:

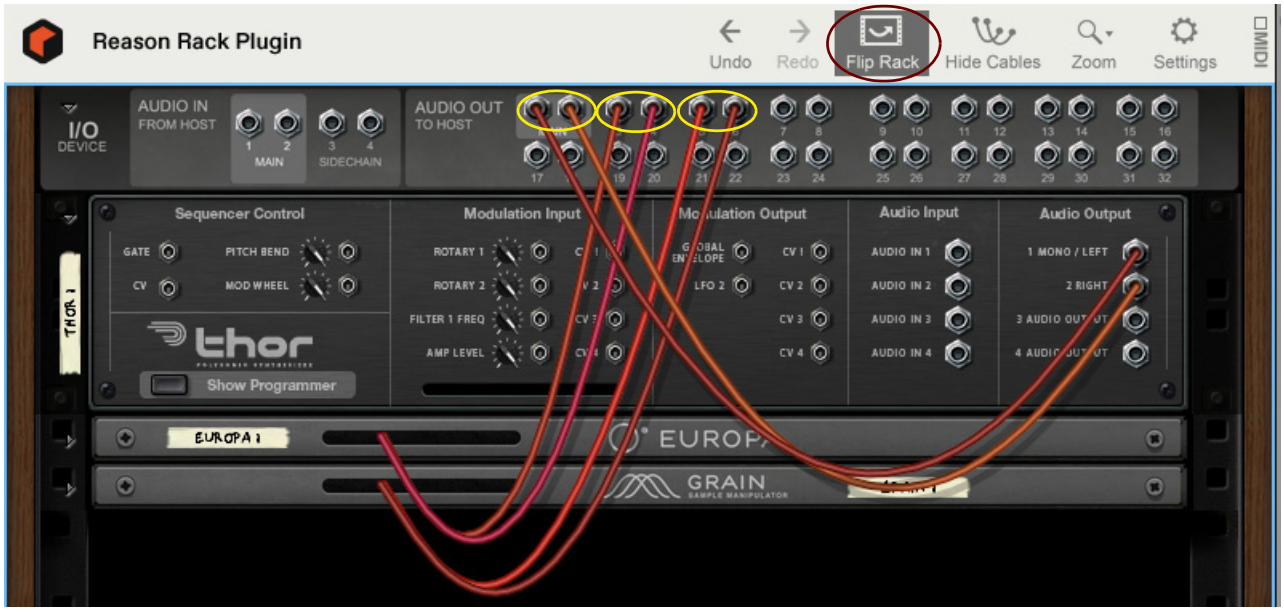


Adding more effects will connect them in series. You can always click Flip Rack to see how devices are connected on the back side of the rack. Read more in the "Routing Audio and CV" chapter about how to do manual routing of signals, for more complex effect setups!

Layering instruments

Layering two or more instrument devices is a quick way to create thicker or more complex sounds. To do this in Reason Rack Plugin, simply add another device below the first one (either by clicking Add Device at the bottom of the rack or by using the Browser).

The new instrument devices will be automatically routed to the first free output on the I/O device at the top of the rack:



By default, sum To Main is activated, which means the instruments will be mixed with the first one on the main stereo output to the DAW:



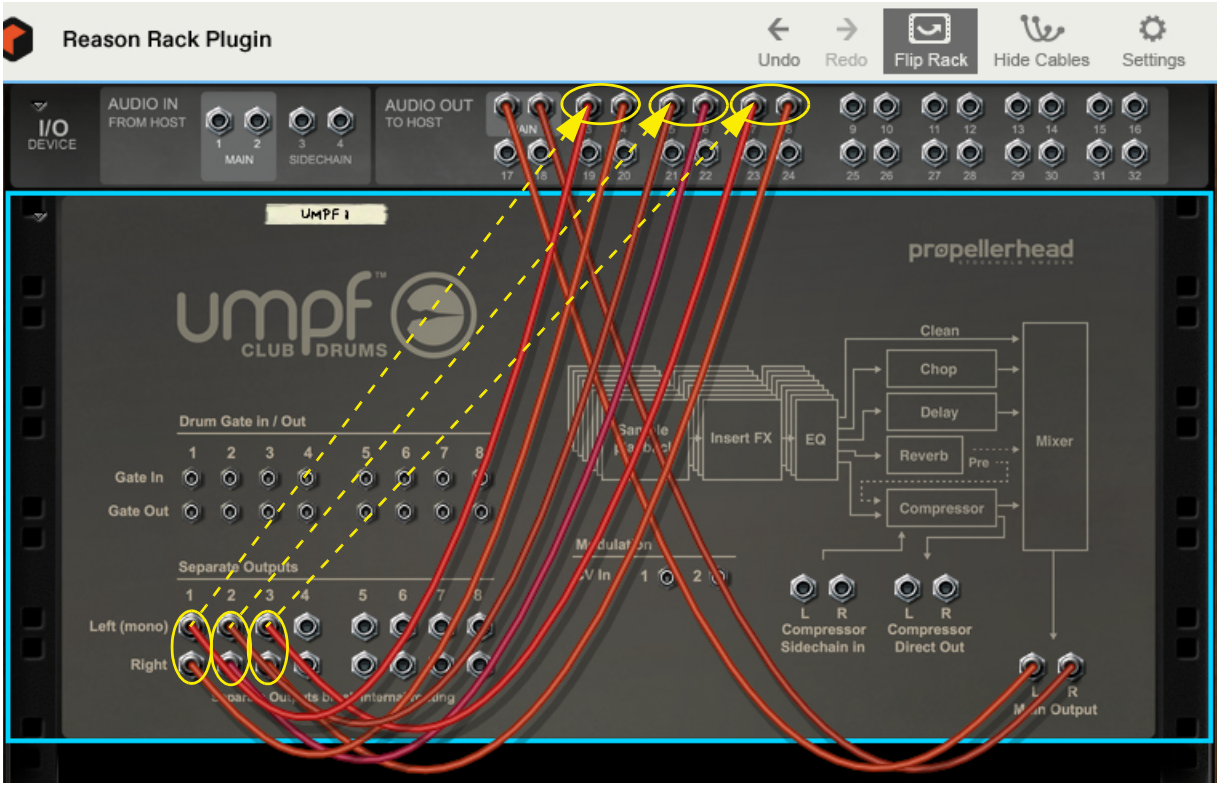
All instruments in the rack will get the same MIDI input, i.e. when you play your MIDI keyboard, all instruments will receive notes and be heard at the same time. For more advanced layering techniques, you can use the Combinator device (see "The Combinator").

Using separate audio outputs

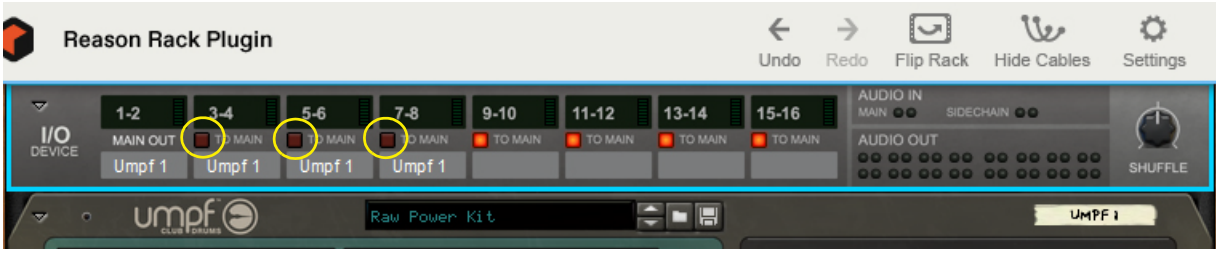
If you have layered instruments as described above but want to send them to separate mixer channels in your DAW host, simply turn off sum “To Main” for their outputs on the I/O device. This will send each instrument device to a separate output, and your DAW will be able to receive them on individual channels.

Some instrument devices have multiple outputs themselves. For example, a drum module may have separate outputs for each drum. To route these to individual channels in your DAW host:

1. **Click the Flip Rack button to show the backside of the rack.**
2. **If the I/O device is folded, click its arrow button to unfold it so you can see the output jacks.**
3. **Scroll down to the instrument device.**
4. **Click on a separate output jack on the device, and drag up to the desired output jack on the I/O device.**
 A cable is shown when you drag.
5. **Release the cable on the jack.**
 It is now connected. If the output was in stereo and you dragged the left channel, the right channel is automatically connected as well.
6. **Repeat the connection procedure for some more separate output pairs:**



7. **Flip the rack around again and make sure sum To Main is turned Off for the separate output pairs.**
 This sends the signals to the DAW on a separate output channels instead of summed to the main output:

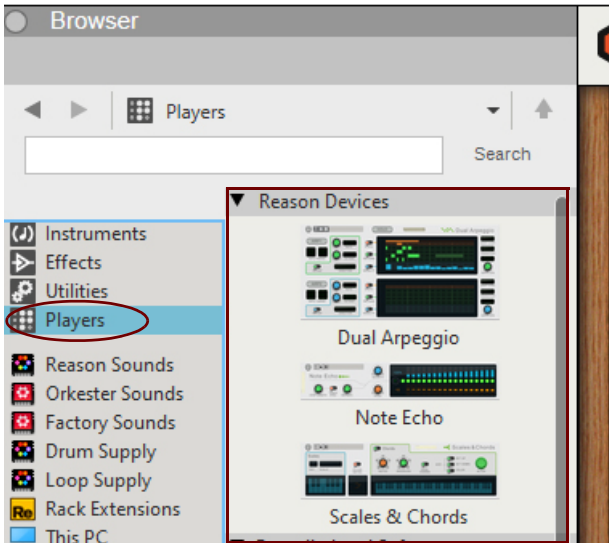


Depending on the instrument, you may also need to make settings on the device itself to assign sounds to that separate output etc. See the documentation for the instrument device.

Adding Players

Players are special devices that transforms or generates MIDI notes and passes them on to an instrument device. A Player can for example create chords or arpeggios from the notes you play, or generate notes without input, like a sequencer or drum machine.

Players are added from the Browser or from the context menu. They are listed on a separate palette or submenu.



A Player is always added to an instrument device - typically you first select the instrument and then add the Player. In the rack, it sits on top of the instrument device, intercepting incoming MIDI notes and passing them on, transformed in various ways. You can also chain/stack multiple Players:



A Scales & Chords Player in series with a Note Echo Player, controlling an NN-XT sampler instrument.

Most Players have On buttons. When turned off, they will bypass MIDI as if they were not connected at all. There is also a Bypass All button at the top - this bypasses the whole chain of Players.

Read more about the included Player devices in "Working with Players".

! **Note that Reason Rack Plugin doesn't send MIDI back to the DAW host. You cannot use a Player as a general MIDI effect for controlling another instrument plugin in the project.**

Using Mixer devices

Although the sum To Main function makes it easy to layer several instrument devices, you may want more control over how they are mixed. This is best achieved by adding a Mixer device in the rack before you add your instruments:

1. **In an empty rack, click the "Add Other Device" button and select the Utilities submenu.**
2. **Under Reason Devices, you'll find the Mixer 14:2 and Line Mixer 6:2 - select one of these to add it in the rack.**



The output of the Line Mixer 6:2 is automatically connected to the I/O device.

3. **When you now add instrument devices, they will be routed to the Mixer inputs, instead of to the I/O device:**



An ID8 instrument and a Monotone instrument added to the rack and auto-routed to the 6:2 Line Mixer.

A Mixer allows you to balance levels of the instruments, mute or solo them and use the Pan controls to place different instruments in different parts of the stereo image. You can also add Send Effects, and the Mixer 14:2 has a basic EQ. Read more about the mixer devices in "Mixer 14:2" and "The Line Mixer 6:2".

Detailed control over MIDI note input

In Reason Rack Plugin, all instrument devices receive MIDI notes (as well as a few effect devices and utilities such as the Combinator and RPG-8). However, sometimes you may not want all devices to receive MIDI notes, or you may want to send MIDI notes to effect devices which normally don't receive them.

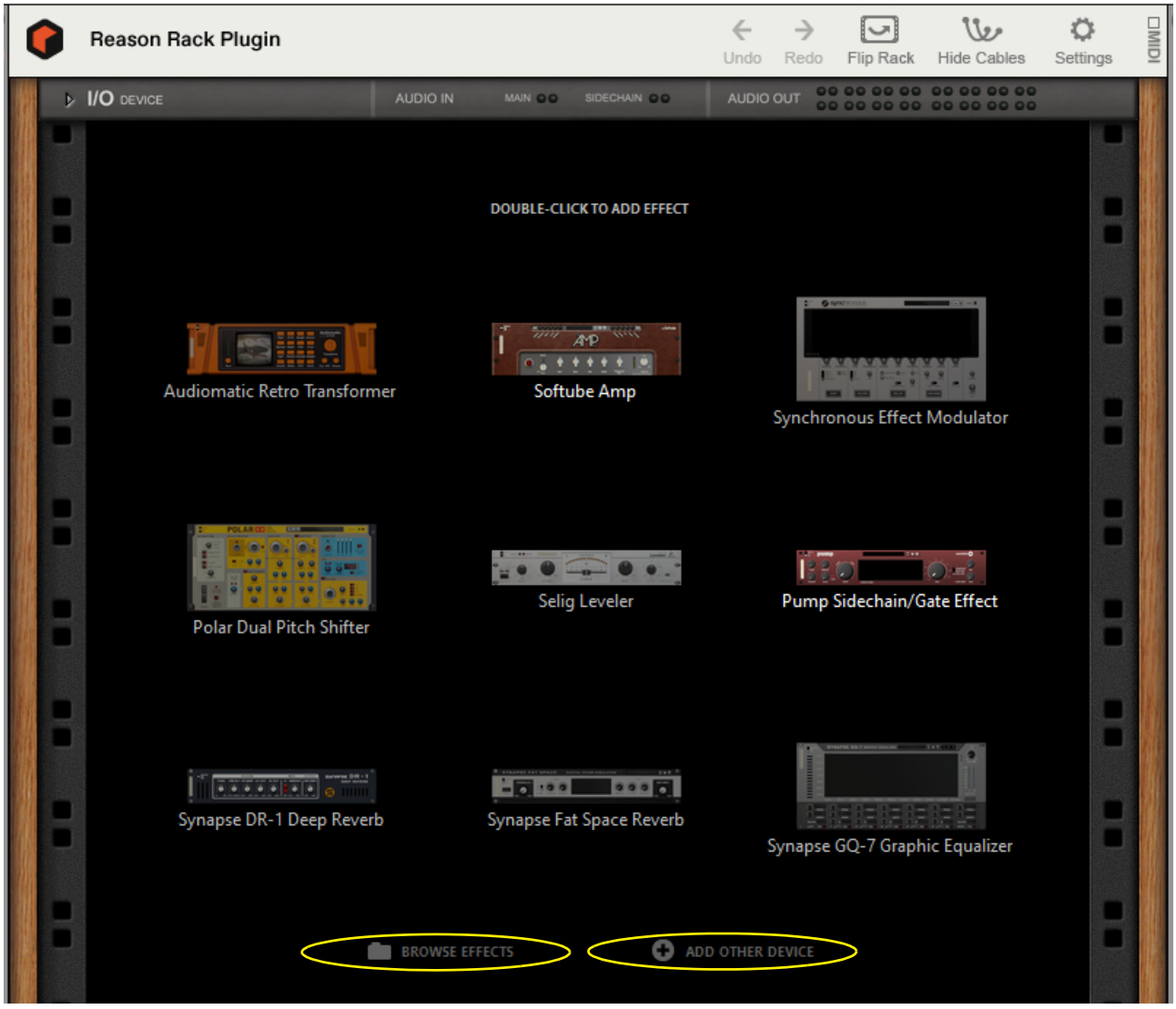
The solution is to select the devices and select "Combine" from the context menu. This puts the selected devices inside a Combinator. Devices inside a Combinator don't receive MIDI directly from the DAW host - instead, the Combinator receives MIDI notes and distributes them to the devices within. You can use the "Receive Notes" setting in the Combi Programmer section to control this for each device. This section also allows you to create keyboard and velocity splits. Read more about the Combinator in "The Combinator"!

The same is true for performance controls such as Mod Wheel, Pitch Bend, Aftertouch and Sustain Pedal. They are normally only received by devices that receive notes, but there are separate settings in the Combi Programmer for these controls as well.

- **By creating a mixer device inside the Combinator and mixing instruments there, you create a self-contained multi-instrument, which can be saved as a combi patch and opened in the standalone version of Reason.**

Chapter 5
Using Reason Rack Plugin as an Effect

Creating an effect



When you add Reason Rack Plugin as an effect the plugin window opens. When the rack is empty, an overlay is shown with icons of the most popular effect devices. Either:

- **Double click an icon to add that effect,**
- **click "Browse Effects" to open the Browser with the Effects palette shown,**
- **or click "Add other device" to add another device from the context menu that appears.**

When you have added a device, the popular devices overlay goes away - it is only shown when the rack is empty. Note that the list of popular devices will change over time to include the instrument devices you most often add!

The first effect device you add will be automatically connected between the Main Input jacks on the I/O device and the Main Output jacks. Audio sent from the DAW host will pass through the effect and be sent back to the DAW:



A The Echo effect device automatically connected to Main In (1-2) and Main Out (1-2) of the I/O device.

When nothing at all is connected to the I/O device, Reason Rack Plugin Effect will send any incoming audio back to the DAW host (as if the rack was bypassed). This way you can still hear your audio when you add an empty rack.

Browsing for patches

Most effect devices come with patches for quickly changing sounds. There are two main ways to select a patch for a device:

- **Use the patch name display on the device panel.**
 Either click and select a patch from the menu or use the previous/next patch buttons next to the display:



- **Click the Browse Patches button on the device (or select Browse Patches from the context menu).**



 This opens the Browser, where you can search and navigate the sound banks and folders on your hard drive. Read more in "Sounds, Patches and the Browser".

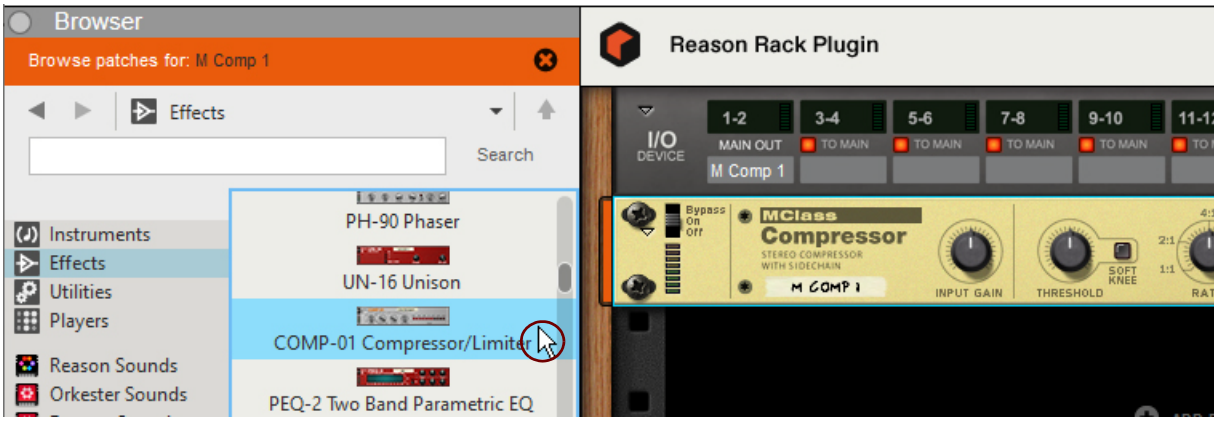
It's also possible to browse between different devices (rather than patches). For example, there are several different Compressor devices available in Reason Rack Plugin, and many of them don't use patches.

1. **Let's say you start by adding an M-Class Compressor to your rack:**

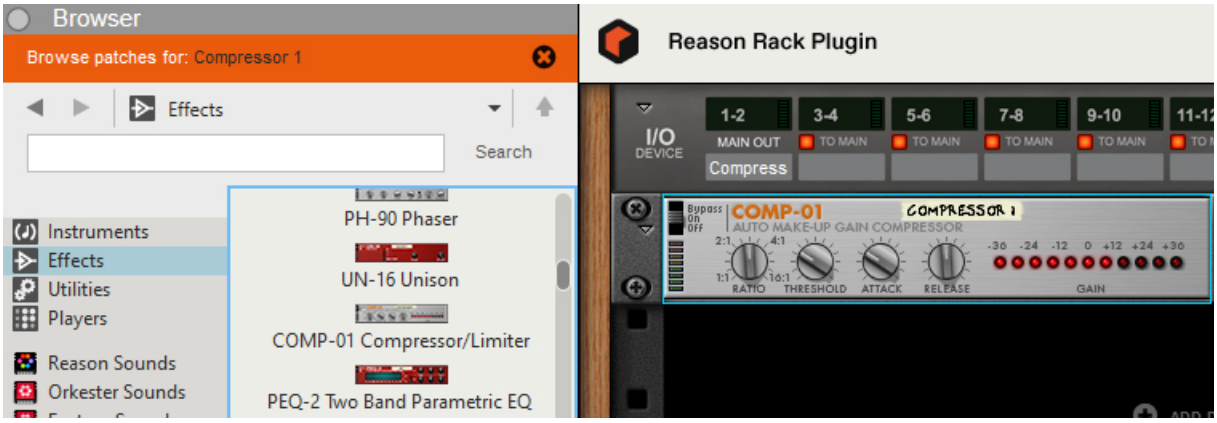


2. **Now, right-click the MClass Compressor device and select "Browse Effects" from the context menu.**
 The Browser opens with the Effect palette shown.

3. **Scroll to find the Comp-01 Compressor/Limiter, then double click it:**



It now automatically replaces the M-Class Compressor in the rack:



4. **Continue until you've found the effect that suits you best.**

This type of browsing, where you replace a device with another device type, is called cross-browsing. You can also use drag and drop from the Browser to do this.

Creating effect chains

- **Adding more effects will automatically connect them in series.**
- **If you press [Shift] and drag an effect up or down in the rack, it will be automatically re-routed, changing the order of effects in the signal chain.**
 Note that this can change the sound drastically!

It is also possible to do more complex setups such as splits and parallel effect chains. For this you could use devices such as the Spider Audio Merger & Splitter (see "Spider Audio Merger & Splitter"). This would require you to do some manual routing by dragging cables on the back of the rack. Read more in "Manual routing".

All effect devices in Reason Rack Plugin have a Bypass/On/Off switch:



This is normally set to On, but setting it to Bypass lets you temporarily disconnect the effect. Setting an effect to Off will silence it completely (no sound will be passed through). This is mainly useful if you are using send effects or parallel effect chains.

Once you have an effect chain that you're happy with, you could Combine it (by selecting all devices and selecting Combine from the context menu). This creates a Combinator with all devices. You can save this as a combi patch and load it into other instances of Reason Rack Plugin or the standalone version of Reason, see "The Combinator".

Using sidechain inputs

Reason Rack Plugin has four audio input channels: Main left + right and Sidechain left + right. This allows you to send an additional stereo audio signal into the rack (provided that your DAW host supports this). How you use this is really up to you, but a common usage would be sidechaining a compressor:

1. **Open Reason Rack Plugin as an effect.**
 For this example, you should add it as an insert effect on a synth pad track or similar.
2. **In the rack, add a compressor device with a sidechain input, e.g. the M-Class Compressor.**
3. **Flip the rack.**
4. **Click the Sidechain input on the Compressor and drag to connect it to the Sidechain In on the I/O device at the top of the rack.**
 (If the I/O device is folded, just hold the cable over the folded device for a moment - it will unfold automatically to allow you to connect it.)



Connecting the Sidechain inputs of the MClass Compressor.

5. **In your DAW host, route another channel or audio track to Reason Rack Plugin input 3-4 (Sidechain input).**
 Typically, you would use something like a kick drum loop for this.
6. **Start playback of your synth pad and kick drum tracks.**
 The synth pad will be processed by the Compressor, but the Compressor will be triggered by the sidechain signal.
7. **Adjust the settings on the M-Class Compressor to get a classic, pumping sidechain pad.**



Chapter 6
Working in the Rack

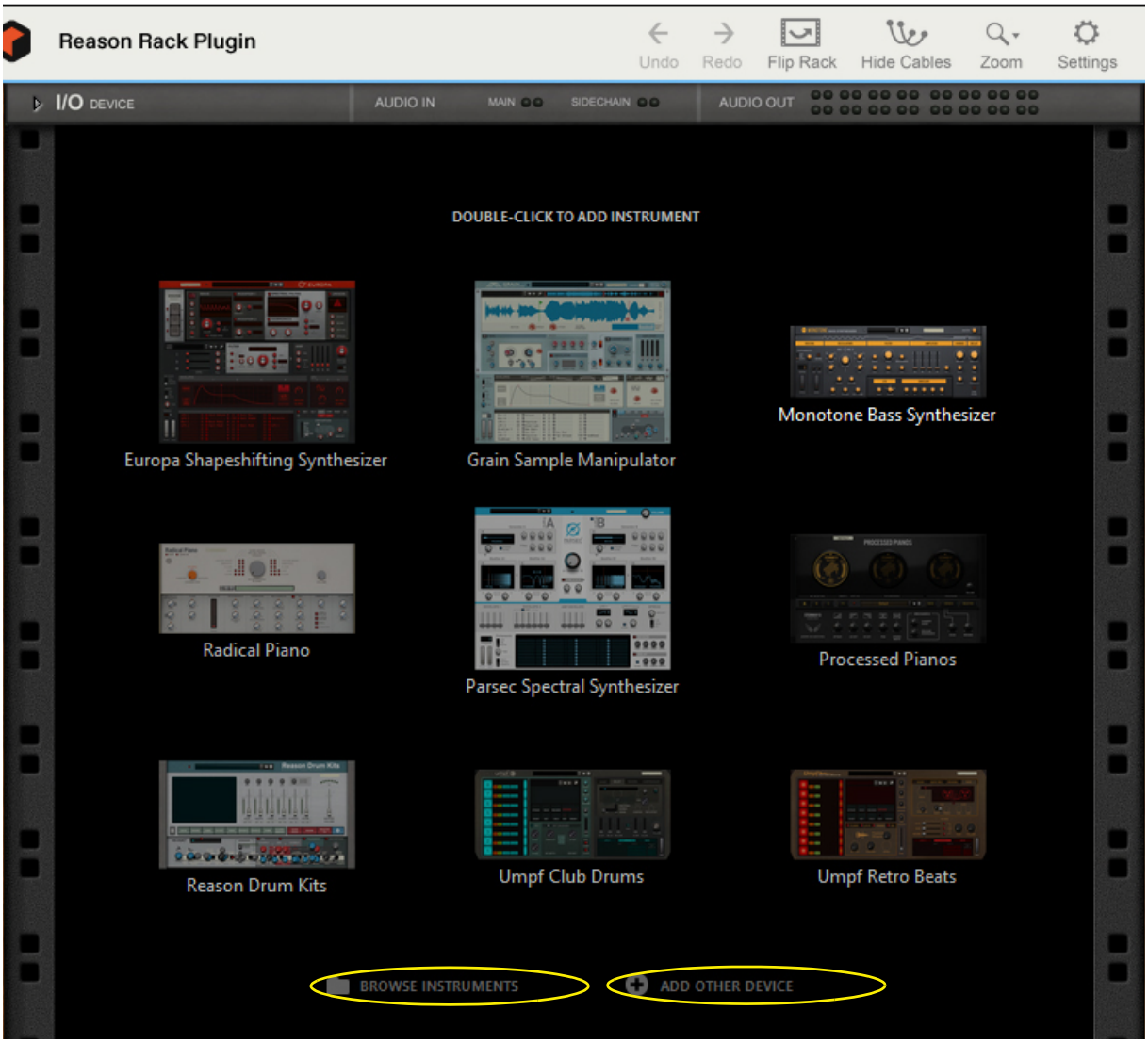


Creating devices

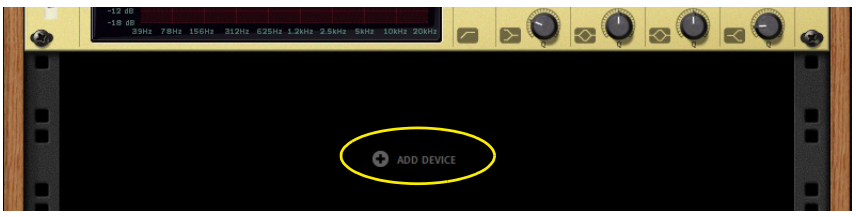
Devices can be created in a number of different ways.

Either:

- **Double click an instrument icon to add that device,**
- **click "Browse Instruments" or “Browse Effects” to open the Browser with the corresponding palette shown,**
- **or click "Add other device" to add another device from the context menu that appears.**



- **If there already are devices in the rack, click the Add Device button (or context-click in the rack) and select a device from the menu:**



→ **Select a device or patch in the Browser and click Create (or double clicking it the device).**

• **Drag and drop a device or patch from the Browser to the Rack.**
As you drag a device or patch to the rack, a +-sign is shown together with an orange divider to indicate where the device will be placed:



Adding a PX7 FM Synthesizer device by dragging from the Instruments device palette and dropping in the rack.

! **When using drag and drop, pay attention to where you drop the device:**

• **If you drop a device on top of an existing device in the rack, you will replace it.**

• **Dropping a patch on top of a device means loading the patch (and possibly replacing the device), while dropping in the empty rack or below devices means creating a new device with that patch loaded.**

Selecting devices

→ **To select a single device, click on it in the rack.**
The selected device is displayed with a colored border (based on the color scheme selected for your operating system).



→ **To select several devices, hold down [Ctrl](Win) or [Cmd](Mac) and click on the desired devices.**

→ **Hold down [Shift] and click to make a continuous (range) selection.**

→ **To de-select all devices, click in the empty part of the rack.**

Moving devices

A device can be dragged freely up and down in the rack without affecting the routings.

In this example an RV-7 reverb device is moved to two different positions:

In this case, the line indicates that the RV-7 reverb device will be placed to the left of the phaser.



This is the result. Note that the filter device is moved to the left, to fill out the gap.



In this case, the line indicates that the reverb device will be placed to the right of the chorus/ flanger.



This is the result. All three devices are moved to the left, to fill out the gap.



Re-routing devices

→ **If you hold [Shift] and drag a device to a new position in the rack (as described above), it will be re-routed (as if you deleted it and created it in its new position).**
This allows you to e.g. change the order of effect devices in a signal chain by Shift-dragging them.

See "Auto-routing" for more info on auto-routing.

Deleting devices

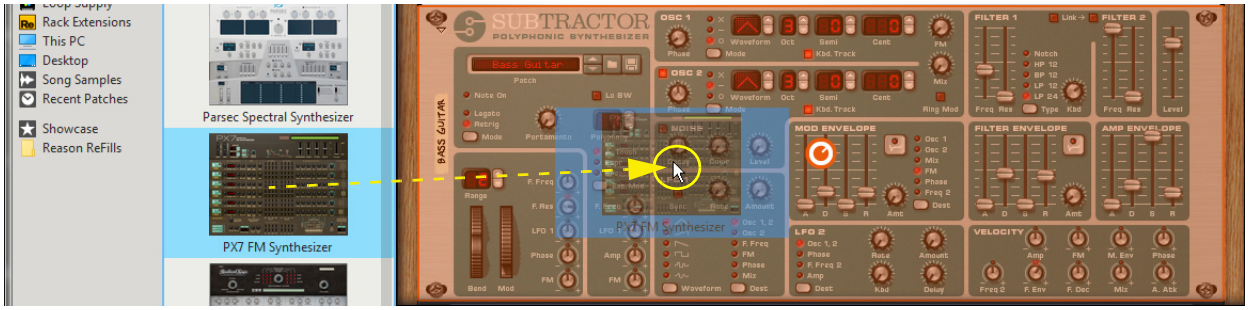
→ **To delete a device, right-click it and select Delete.**
If the deleted device was part of a chain, the signal chain will be kept.

- **You can also select multiple devices and then select Delete to remove them all in one go.**
- **Deleting all devices in the rack will make the Popular Devices palette show.**

! **The I/O device is fixed to the top of the rack and cannot be deleted.**

Replacing devices

→ **Drag and Drop a device on top of an existing device in the rack, to replace it.**
When dragging a device on top of another device in the rack, the panel of the existing device is shaded in orange:



Replacing a Subtractor device with a PX7 FM Synthesizer device by dragging from the Instruments device palette and dropping on the Subtractor.

Cut, Copy, Paste and Duplicate devices

These functions on the context menu affect the currently selected devices.

← **Note that you can also copy and paste devices between different Reason Rack Plugin instances in your DAW project!**

Naming devices

Each device has a "tape strip" which shows the name of the device. Normally, this is the name of the loaded patch (or the device type if it doesn't support patches), but you can rename it by clicking the tape strip and typing. This is especially useful if your rack contains several devices of the same type and you need to separate them.



→ **To revert to the default patch name, double click the tape strip and delete your custom name.**

The device name is also shown on the I/O device, if it's connected to one of the first 8 input pairs.

Chapter 7
Routing Audio and CV

Working on the back of the rack

As you've seen in the previous chapters, instrument and effect devices are connected automatically when you create them. This means you don't strictly have to do any manual signal routing to use Reason Rack Plugin, at least not for standard instrument and effect functionality.

However, adding some manual routing vastly opens up the possibilities! Here are some of the things you can do:

- **Connecting multiple outputs from instruments.**
- **Sidechaining effects.**
- **Splitting, merging and mixing signals, creating parallel signal chains, send effect structures and more.**
- **Using CV (Control Voltage - see "CV/Gate signals" below!) to modulate or control parameters on one device from another.**
- **Playing instruments from CV sequencers and arpeggiators.**
- **Using instrument devices with audio input as effects.**

All this routing is done on the back side of the rack:

→ **Click Flip Rack to get there (or select Flip Rack from the context menu):**



All audio and CV connections are represented by cables.

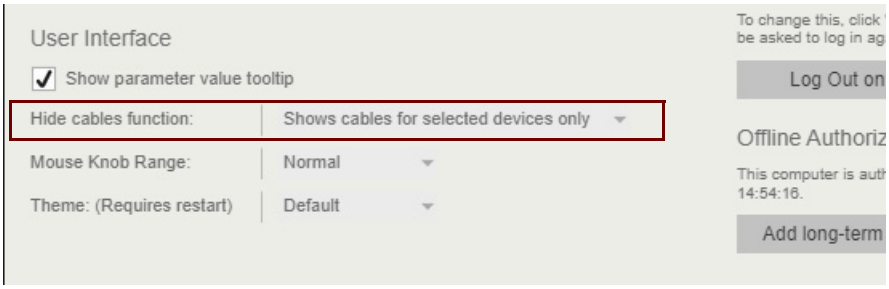
Hiding cables

Sometimes, there can be quite a lot of cables in view, making the connections hard to follow. Then the Hide Cables function is handy:

→ **Click Hide Cables on the Global Panel to toggle this on or off (or select Hide Cables from the context menu).**



The result of Hide Cables depends on this setting in the Settings dialog:



Option	Result
Hides auto-routed cables.	Only cables you have connected manually will be fully shown. Cables that were connected automatically are drawn semi-transparent.
Shows cables for selected devices only.	Only cables connected to the currently selected device(s) are fully shown. Other cables are drawn semi-transparent.
Hides all cables.	All cable connections are indicated with colored dots in the jacks, and no cables will be shown.

Signal types

The following signal types are used in Reason Rack Plugin:

Audio signals

Audio means sound being sent from one device to another (or to/from your DAW host).

- **Audio connectors are shown as large quarter inch jacks and the cables are thick:**



- **Audio connectors can be either inputs or outputs, as indicated on the panel.**
- **You always route cables between inputs and outputs (it doesn't matter in which order you route).**
 The special case is the i/o device at the top of the rack, where the "Inputs" represent the inputs from the DAW host and the "Outputs" represent the outputs to the DAW host. Thus, an instrument will have its own Output jacks connected to the "Outputs" on the i/o device, to get the sound back into your DAW host.
- **Many devices have audio jacks in stereo pairs, with one labeled e.g. "Left/Mono" and the other "Right".**
 When you connect such a Left output, the Right is typically automatically connected too - but you disconnect it to use a mono signal from the device.

CV/Gate signals

In the early days of synthesizers, before the MIDI protocol was invented, analog synthesizers could be interconnected using Control Voltage (CV) cables. For example, one cable would be used for controlling pitch while another would send a Gate voltage, basically telling a synth when to play a note and when to stop. A third cable might send a modulating signal to some parameter, e.g. varying the filter frequency. Today, this system has become quite common again, thanks to the rise of modular synthesizers.

The CV signal cables in Reason Rack Plugin emulate this analog control system. CV cables send a value, which may be static or changing. They do not carry audio, but are used for modulating parameters and controlling devices.

- **CV connectors are shown as smaller mini jacks, and the cables are thinner than the audio cables:**



- **CV connectors can be either inputs or outputs, as indicated on the panel.**
 You always route cables between inputs and outputs (it doesn't matter in which order you route).

- **Connectors may be labelled "Gate".**
 A Gate signal is a CV signal that goes from zero to an "on" value and eventually back to zero again. They are used for playing notes, triggering envelopes and more.
- **Instrument devices often have a "Note" CV input, along with a "Gate" input.**
 These are used in tandem, to play the instrument from a CV source such as the Matrix rack sequencer. The Gate signal will determine when notes start and stop, while the Note CV signal will set the pitch of the played notes.

About CV Trim knobs



Most CV inputs have an associated Trim knob. This is used to set the CV "sensitivity" when modulating a parameter. The further clockwise a CV trim knob is set, the more pronounced the modulation effect.

- **Turned fully clockwise, the modulation range will be 100% of the parameter range.**
- **Turned fully anti-clockwise, no CV modulation will be applied.**

About MIDI routing

MIDI signals (notes and controllers such as pitch bend, mod wheel etc) are not represented by cables in Reason Rack Plugin. Normally, all devices that use MIDI notes and performance controllers will automatically receive them from the DAW host.

- **To decide exactly which devices should receive MIDI notes, put them in a Combinator and use the "Receive Notes" setting in the Combi Programmer section for each device. See "Configuring the Combinator panel".**
- **If a Player is added to an Instrument device, MIDI notes will go to the Player instead of the instrument.**
 The Player may pass on the notes, transform them or generate new notes, depending on the Player type and settings.
- **You can also control individual parameters with MIDI controller messages (e.g. if you have a MIDI keyboard with knobs and faders).**

Manual routing

There are two ways to manually connect an output jack to an input jack (or vice versa):

Dragging cables

1. **Click the jack, and keep the mouse button pressed.**
 A cable appears.
2. **Drag the cable to the other jack.**
 When you're over a jack of the correct type, it lights up.
3. **Release the mouse button.**
 The cable is connected (replacing the existing connection there, if any). If you dragged from the left jack in a stereo pair, the right will automatically be connected as well.

→ **To change a connection, click and hold the jack to grab the cable. Then drag it to another jack.**

→ **To disconnect a cable, click the jack at either end to grab the cable and drop it away from any jack.**

Using the routing menu

1. **Right-click a jack.**
 A context menu appears, listing all devices in the rack:



2. **Move the mouse pointer to the device you want to connect to.**
 A submenu lists all outputs or inputs on that device. An asterisk (*) next to a jack means it's already connected.
3. **Select the desired jack.**
 The two jacks are connected with a cable. If the jack was already in use, the old connection is replaced.

→ **To disconnect a jack, right-click it and select "Disconnect" from the context menu:**



Checking and following cable connections

→ **If you point your mouse at a jack that's in use and wait a moment, a tool tip appears, showing both the device name and connector that the cable is connected to:**



→ **You can also right-click the jack and select "Scroll to Connected Device".**
This will scroll to the device in the other end of the cable and highlight the connector briefly.

Auto-routing

Reason Rack Plugin will automatically route devices when you create them. If you don't want this, you can hold down [Shift] when you create the device. This will add the device without connections, requiring that you route it manually to use it.

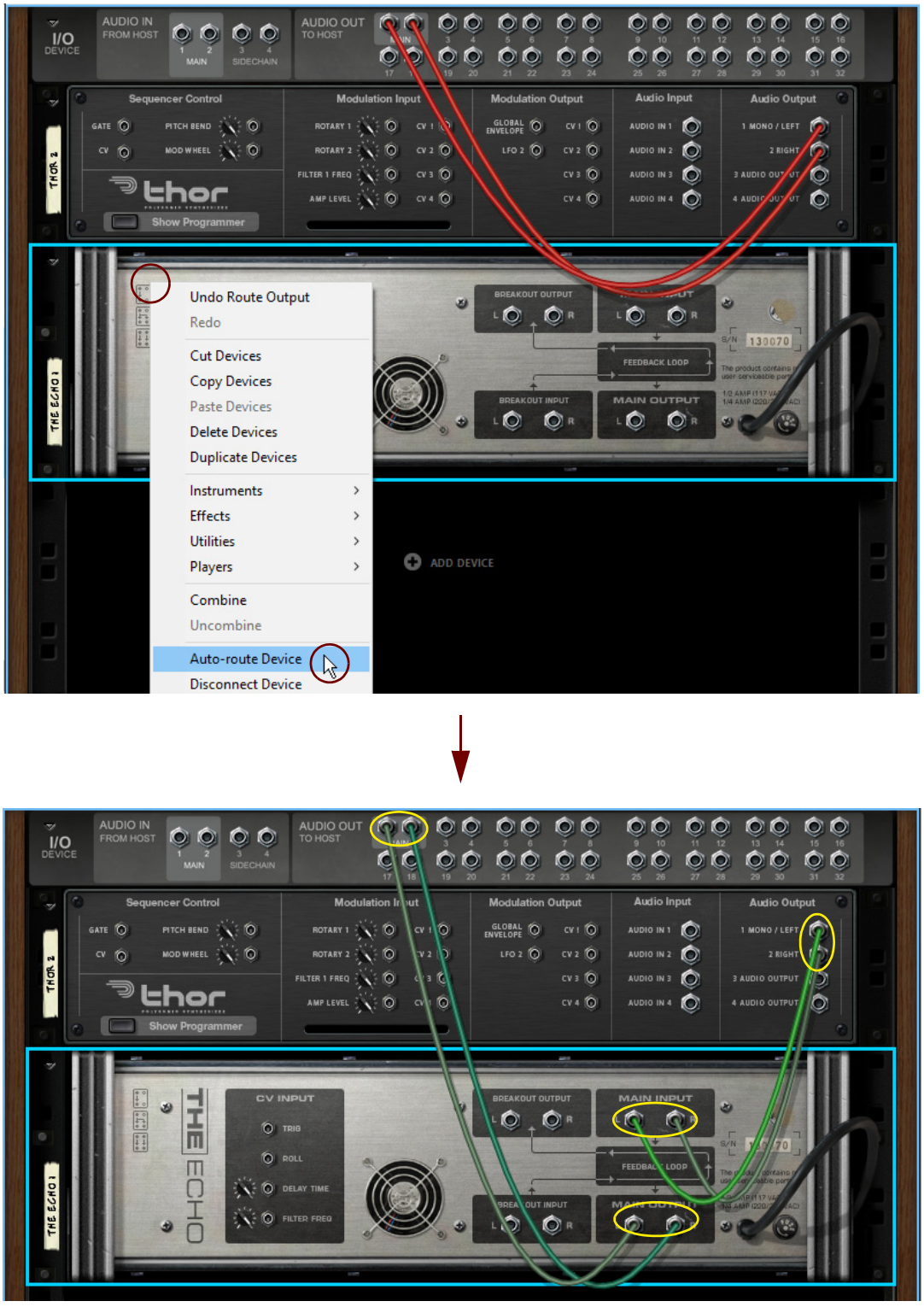
It's also possible to invoke this automatic routing from the context menu, by right-clicking a non-connected device and selecting "Auto-route Device".

To disconnect all cables going to and from a device, instead select "Disconnect Device" from the context menu.

When a device is auto-routed, either at creation or from the context menu, the following rules apply:

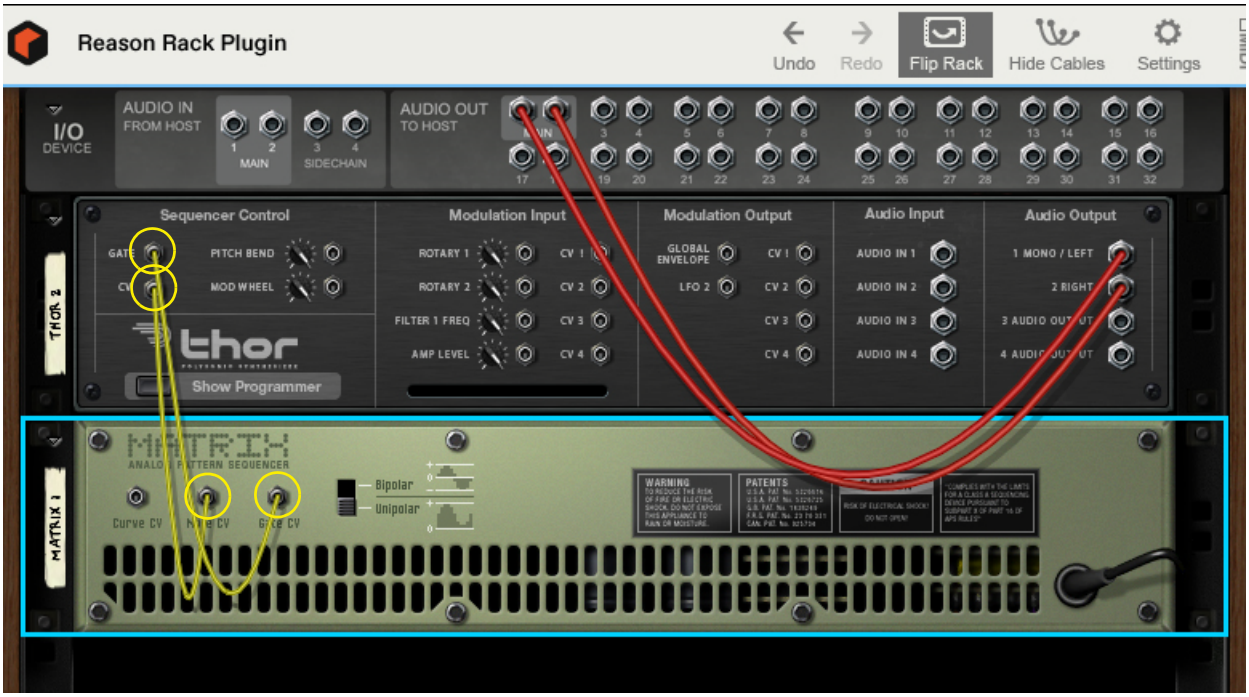
- **An audio output will be connected to the first free and auto-routable input above it in the rack.**
 For example, if you auto-route an instrument and there's a rack mixer device above it in the rack, it will be connected to the first free mixer channel input. If there are no such free, suitable inputs on devices above, it will be routed to the I/O device instead.

- **An audio input will be connected to the first auto-routable output above it in the rack.**
 If this was already connected, the device will insert itself into the signal chain but preserve the connection.



Auto-routing a non-connected The Echo device below a Thor instrument device.

- **Devices inside a Combinator will not auto-route outside the Combinator device.**
- **A few CV devices will auto-route Note CV and Gate cables to the corresponding CV inputs on the first suitable instrument device above in the rack.**
 For example, if you add a Matrix sequencer below a Thor synth, it will auto-route Note CV and Gate and be ready for immediate playback:



Chapter 8
Sounds, Patches and the Browser

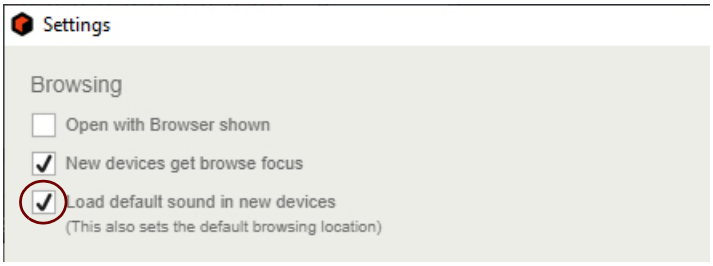


About patches

A patch contains settings for a specific device. Patches can be either separate files on your hard disk or files embedded in a ReFill (see "About ReFills" for info about ReFills). A Rack Extension often comes with patches embedded within the Rack Extension itself.

- **A patch most often includes all parameter settings on the front panel, but not cables and trim pot settings on the back side.**
 However, a few Rack Extensions have additional parameters on the back panel which are included in the patches.
- **A Combinator patch is special in that it contains both settings for the device itself (the Combinator settings) and all settings for all contained devices.**
 This includes all routing within the combi, and all settings on the back panels as well.
- **Patches for devices that use samples (samplers, loop players, drum machines) contain references to sample files on disk.**
 The samples themselves are not part of the patches.
- **Some devices don't have patch support at all.**
 Should you need to save the state of such a device, put it inside a Combinator device and save a combi patch!

About the "Load Default Sound in New Devices" setting



If this is activated in the Settings dialog, a default patch is loaded when a device is created. This way, the device is ready for playing right away. This will also determine the default folder when you browse for patches for the device.

If you turn this setting off, new devices will be initialized - parameters are reset to their default values and no samples are loaded in sample-based devices.

Loading patches

Loading patches can be done in the following ways:

- **By using the patch selector directly on the device panel.**
 Either click and select from the menu that appears, or use the Previous/Next Patch buttons next to the display.
- **By opening the browser, dragging a patch from there and dropping it on a device.**
- **By dragging and dropping patch files from a Windows / Mac window and dropping it on a device.**
- **By clicking the Browse Patch button for a device or selecting Browse from the context menu.**
 This opens the browser and gives the device "browse focus" - read more about this in "Opening the Browser and setting Browse Focus" below.

Saving patches

! **All settings for all devices in your Reason Rack Plugin instances are automatically included when you save the song or project in your DAW host - you don't have to save the patches separately!**

Saving patches is useful if you have an instrument or effect setup that you want to use later, in other projects.

Saving is always done by clicking the Save Patch button on the device panel and specifying a name and location for the file in the dialog that appears:

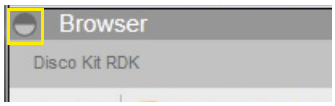


The Save Patch button on a device.

- **If you have modified an existing patch and want to save it with the same name (overwriting the old version), press [Alt](Win) or [Option](Mac) and click the Save Patch button.**
 This only works if the patch was loaded from a file on disk. If the patch was in a ReFill or a Rack Extension, you cannot overwrite it and must save it elsewhere.

Opening the Browser and setting Browse Focus

You open and close the Browser by clicking the circle button in its top left corner:



It is also opened automatically when you set browse focus to a device. You set browse focus in one of the following ways:

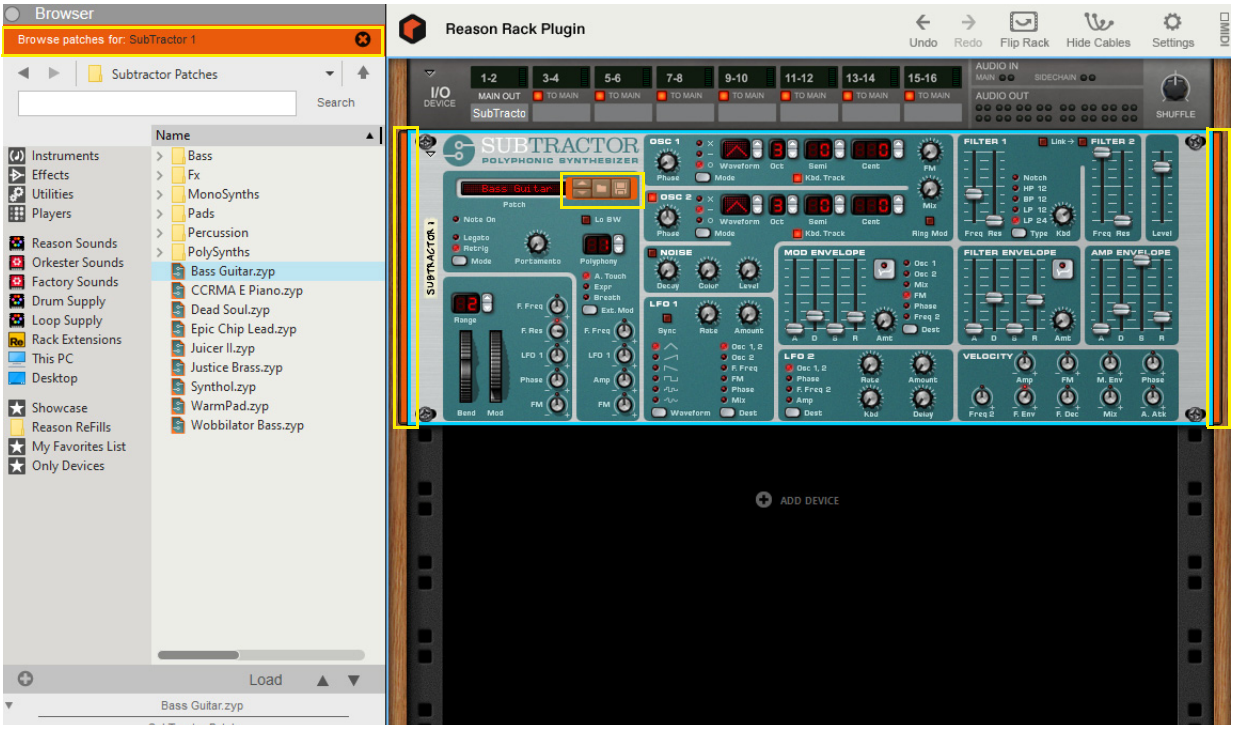
- **By clicking the Browse Patch button on a device:**



- **By selecting Browse from the context menu for a device.**
- **By selecting a device in the rack and clicking the browse button at the top of the Browser:**

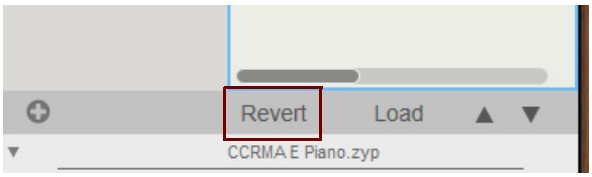


Browse focus is indicated by an orange header in the Browser. The device is shown with orange side bars and an orange-colored patch section:



In this mode, the Browser is "locked" to the device and only displays patches compatible with this particular device type. For example, if you set browse focus to an instrument device such as Europa, only instrument patches for this device will be shown.

- **To load a patch from the Browser in Browse Focus mode, double click the patch in the Browse list or select it and click the Load button at the bottom of the Browse list.**
 You can also step up or down in the list of patches shown with the arrow buttons below the list - this will automatically load the selected patch
- **To go back to the original patch for a device in this mode, click the Revert button at the bottom of the Browse list.**
 Note that the Revert button appears only after you have loaded at least one other patch in the device:



- **When you're happy with the loaded patch, you clear browse focus by clicking the X button at the top of the browser (or by clicking elsewhere in the rack):**



Why can't I see all files?

Sometimes, some patches or samples may seem to be missing from a folder, or a folder might look completely empty. Check if the Browser is in Browse Focus mode - is the header orange? Remember, in this mode, the Browser will not show all items, only those that match the selected device. This may mean that only instrument patches or effect patches are shown, or only loops or samples!

→ **Click the X button to leave Browse Focus mode and see all items again.**



Browser settings

In the Settings dialog are two options for the Browser:

Option	Result
Open with Browser Shown	By default, the Browser area is folded when you open Reason Rack Plugin. Tick this option if you want it to be shown right away.
New devices get browse focus	If this is on when you create a device (and the Browser area is shown), the device will get browse focus, as if you had clicked its browse button. This means that the Browser will automatically go to the default folder for the device and show the right device type (Instrument, Effect, Utility or Player).

About cross-browsing

When you are browsing patches for an instrument device, you are free to load any kind of instrument patch. For example, if you're browsing patches for a Europa synthesizer, you can load a Radical Piano patch. This will replace the Europa device with a Radical Piano device and load the patch.

Cross-browsing is useful in several ways:

- **When you're looking for a sound of a certain type (e.g. a synth bass), you don't have to care about which device to select - you can just search in the Browser and step through the list of patches in the search result.**
- **You can make Favorite lists with different patches and step through these freely (see "Favorites and Favorite Lists").**
- **By using the Instruments and Effects palettes at the top of the browser, you can even cross-browse between devices that don't use patches.**

! **You can cross-browse Instruments, Effect devices or Players, but you can only do it within category (it isn't possible to replace an effect with an instrument, etc).**

! **Cross-browsing isn't available for Utility devices.**

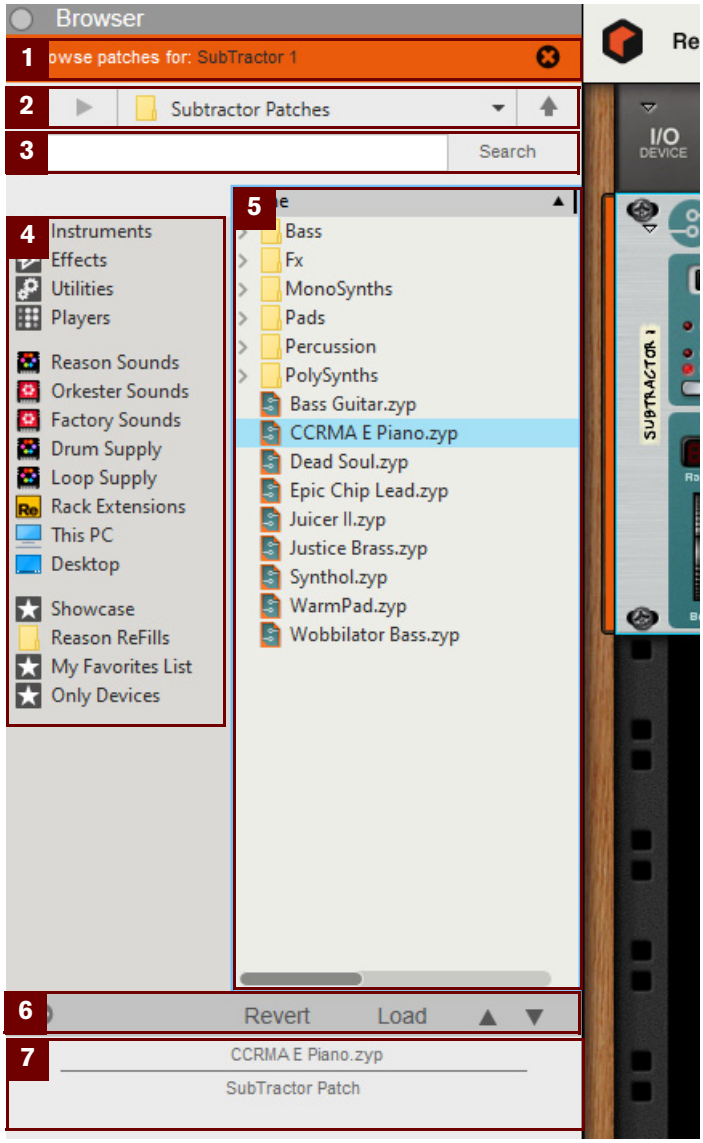
Special instances of cross-browsing

There are a few instances when cross-browsing between device types might lead to lost cable connections in the rack:

- **Non-standard audio connections may be lost.**
 An example is replacing an NN-XT, which can use up to 16 audio outputs, with a Subtractor which only has one audio output.
- **CV connections on the back panel may be lost.**
 The only connections that are retained between device types are Sequencer Control CV/Gate in.

! **If you encounter such situations and you want to restore the original connections, use the "Undo" function. Browsing back to the original device patch will not restore lost connections.**

Browser details



The Browser when using the Browse Patch button/function on a Subtractor instrument device.

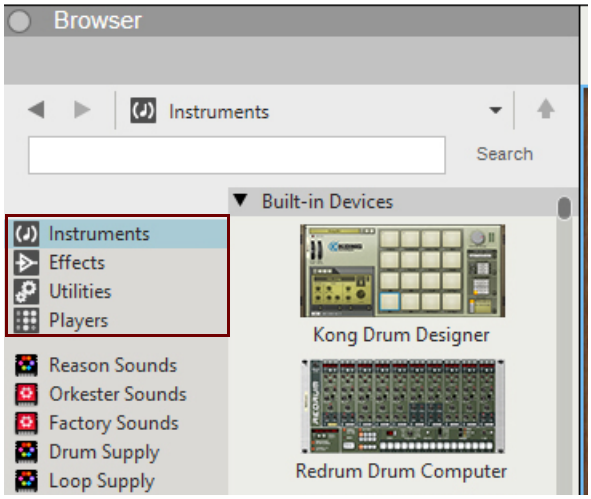
The Browser has the following sections:

- **1. Browse focus field**
 Shown in orange when a device has browse focus. Click the X button to the right to set or clear browse focus.
- **2. Navigation field**
 Shows the name of the current folder. To the left are back and forward buttons, and to the right is an Up button for going to the parent folder.
- **3. Search field**
 Type something and click Search to find patches or samples of that name in the current folder.

- **4. Locations and favorites**
 The column to the left contains shortcuts for going directly to various locations. At the top are the four palettes (Instruments, Effects, Utilities and Players), showing all included devices. In the middle are fixed locations such as the sound banks, and below these you can add your own locations and favorites (see "User Locations and Favorite Lists").
- **5. Browse list**
 Shows the contents of the current folder, with hierarchical subfolders. If you have made a search or selected a favorite list, this will be a flat file list instead. This is where you select the file to load. Scroll to the right to see more columns, for sorting the list.
- **6. Controls**
 The area directly below the browser list holds buttons for Load, Previous/Next and Revert. If you are browsing samples, there are also audition controls here (see "Browsing samples and loops").
- **7. Info area**
 Shows information about the selected item. This area can be folded with the arrow button in the left corner.

► **When you are browsing patches for an instrument device, you are free to load any kind of instrument patch. For example, if you're browsing patches for a Europa synthesizer, you can load a Radical Piano patch. This will replace the Europa with a Radical Piano device.**

The Device Palettes



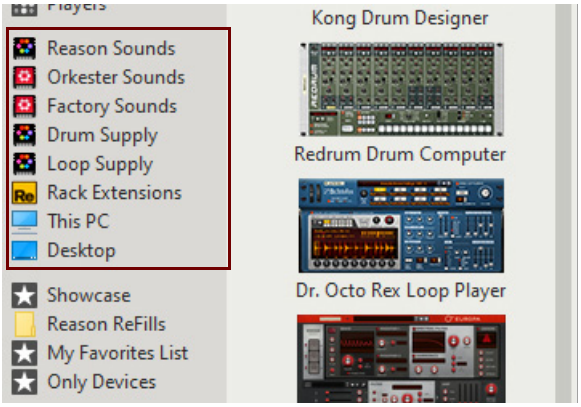
Click the shortcuts at the top of the area to the left in the browser to show one of the device palettes. These show icons of all devices in Reason Rack Plugin. At the top you'll find the built-in devices (under "Built-in Devices"); below you'll find all installed Rack Extensions sorted by manufacturer.

- **You can fold or unfold a section in the palette by clicking the arrow button next to the manufacturer name.**
- **Searching when a device palette is shown will find both device names and manufacturer names.**
 Since most devices have longer names that describe their functionality, you can also search for something like "compressor" or "drum" to quickly get a list of devices of that kind!

→ **To add a device from the palette, use drag and drop, double click it or select it and click the Create button at the bottom of the Browse list.**

! **Note that Players can only be added directly above instrument devices in the rack.**

The Sound Banks and fixed Locations



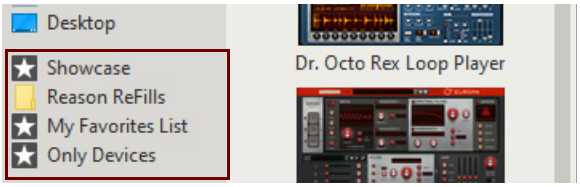
Below the shortcuts to the device palettes you will find the sound banks installed with the program. The sound banks are all ReFills, a kind of component package for Reason which can contain sounds and effect patches, samples, REX files and more:

- **Reason Sounds**
 This is a huge selection of instrument patches, sorted by category. If you for example need an Electric Piano sound, go to the Keys & Chords folder and then into the Electric Piano subfolder for a wide variety of patches for different devices.
- **Orkester Sounds**
 A separate sound bank containing sampled orchestral instruments: strings, woodwinds, brass and more.
- **Factory Sounds**
 Contains patches for all included devices (sorted by device type), samples and REX loops. This is also where you find effect patches.
- **Drum Supply**
 A great collection of modern drum samples and drum kit patches for the Kong Drum Designer. This is an optional download, which can be downloaded from Manage Content on the Window menu in the stand-alone Reason program.
- **Loop Supply**
 Drum and percussion loops in a number of contemporary styles, for use with the Dr Octo REX loop player device or the Nurse REX module in Kong. This is an optional download, which can be downloaded from Manage Content on the Window menu in the stand-alone Reason program.
- **Rack Extensions**
 Most Rack Extensions come with patches that are contained within the Rack Extensions themselves. This virtual location lists all installed Rack Extensions and allows you to browse into them as if they were regular file folders.

 A Rack Extension may also contain patches for other devices as well! For example, they often come Combi patches that combine the Rack Extension with stock effect devices, etc.
- **Fixed folder locations**
 To navigate in the file system, there are shortcuts to your user folder and the desktop. These cannot be removed.

! **If you have un-synced cloud folders (e.g. in Dropbox) in the Locations and Favorites list you might want to exclude these folders from the Browser indexing, to avoid very long indexing times.**
See "Do not index these folders" for more information.

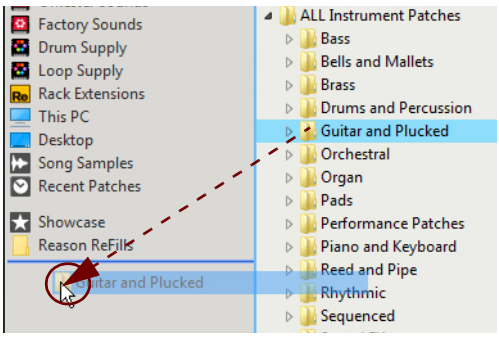
User Locations and Favorite Lists



User Locations

User Locations are shortcuts to folders, either on your hard disk or within a ReFill or Rack Extension. It might for example be the folder where you save your patches, or a particularly useful subfolder in the Factory sound bank.

→ **Add a Location by dragging a folder from the Browse list to the lower part of the left browser section:**



Adding a new Location.

You can then click it to quickly navigate into that folder.

→ **To remove a Location, right-click it and select Delete.**
This will only remove the shortcut, not affect the actual folder or its contents.

If a User Location folder has been removed or renamed on disk, Reason Rack Plugin won't be able to find it. It will be shown with a yellow warning triangle in the list.

! **If you have un-synced cloud folders (e.g. in Dropbox) in the Locations and Favorites list you might want to exclude these folders from the Browser indexing, to avoid very long indexing times.**
See "Do not index these folders" for more information.

Favorites and Favorite Lists

In the same section as User Locations, you can add Favorites:

→ **Drag a patch, loop or sample to the lower part of the left browser section to create an alias to it, for instant access.**
The Favorite can be used like any other file in the browser.

→ **To remove a Favorite, right-click it and select Delete.**
This will only remove the alias, not the actual file.

It's also possible to create Favorite Lists. These are virtual folders where you can collect aliases to files you like or need to use often:

1. **Click the + button below the Favorite section:**



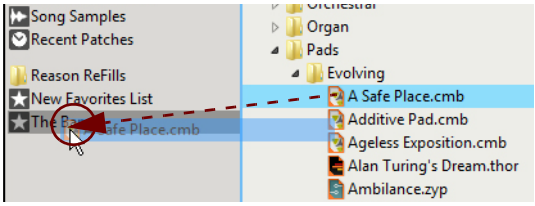
The "Create New Favorites Lists" button

A new Favorite List is created.

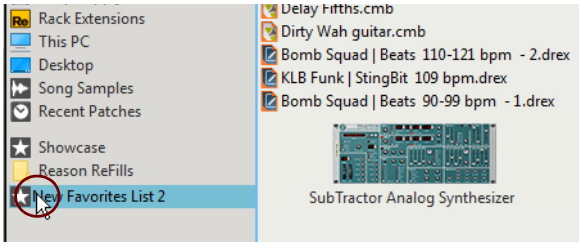
→ **If you like, type in a name for the list.**
You can do this later by double clicking the list and typing.

2. **Click the Favorite List to select it and show its contents.**
At this point, it's empty.

3. **Navigate elsewhere in the Browser, find the desired files and drag them onto the Favorite List.**
Note that you can also add devices from the device palettes:



4. **When done, click the Favorite List to see the items you have added:**



→ **If you like, you can drag them to reorder them.**

← **Note that devices can also be included in Favorite Lists, by dragging them from any of the Palettes!**

→ **To delete a Favorite List, right-click it and select Delete. This will only remove the list and the shortcuts in it, not the actual items.**

Browsing samples and loops

Some devices support loading samples or REX loops (a special audio file format for playing back loops in any tempo). As with patches, you typically can load samples and loops directly from the device panel or from the browser, but there are a couple of things to note:

- **A device may have multiple sample browsers.**
 For example, the Redrum drum machine has ten drum channels, each of which has its own sample browser. When you use drag and drop to load samples, it matters on which drum channel you drop the sample:



Dragging and dropping a sample in Redrum channel 3.

When you click a Browse Samples button, that sets Browse Focus to that particular drum channel on the Redrum device:



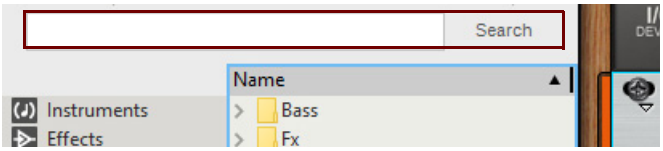
Browsing for a sample in Redrum channel 2.

When a sample or REX file is selected in the Browser, audition controls appear below the browse list:



→ **Click the small triangular Play button to audition the sound - or activate Auto to automatically play back the sounds when you select them.**
The slider to the right sets the audition playback level.

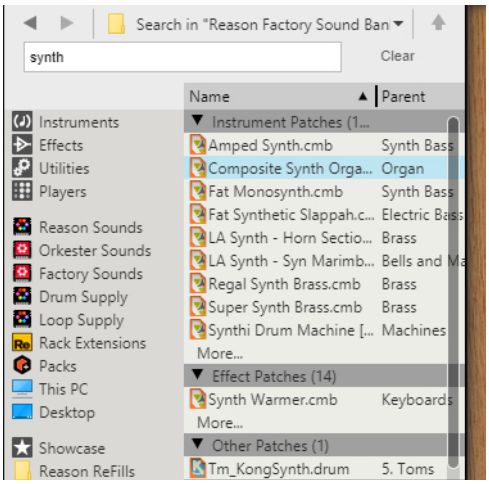
Searching in the Browser



The Search function allows you to search for patches, samples and loops:

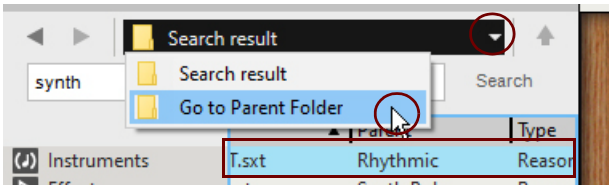
1. **Navigate to the folder, or click a Location in the Browser, in which you want to search.**
 The search will happen in that folder/Location and all its subfolders, so if you e.g. click the Factory Sounds short-cut, you will search the whole factory sound bank.
2. **Click in the Search field, type in a text and click the Search button.**
 The search result is shown in alphabetical order, in separate groups. If you search in the Factory Sound Bank the result is divided into “Instrument Patches”, “Effect Patches”, “Other Patches”, “Samples” and “Folders”, with files and folders whose names match the search string.

 If there are no search hits, the Browser list is empty.

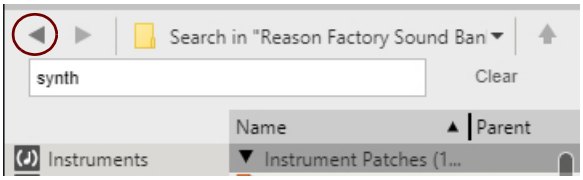


→ **To redo the search, type something else in the Search field and click Search again.**
You don't have to go back to where you started to do this.

If you scroll to the right in the search result list, you'll see a Parent column, showing the parent folders for each item. To go to the folder for an item, select it, click "Search Result" at the top and select "Go to Parent Folder" from the menu that appears:

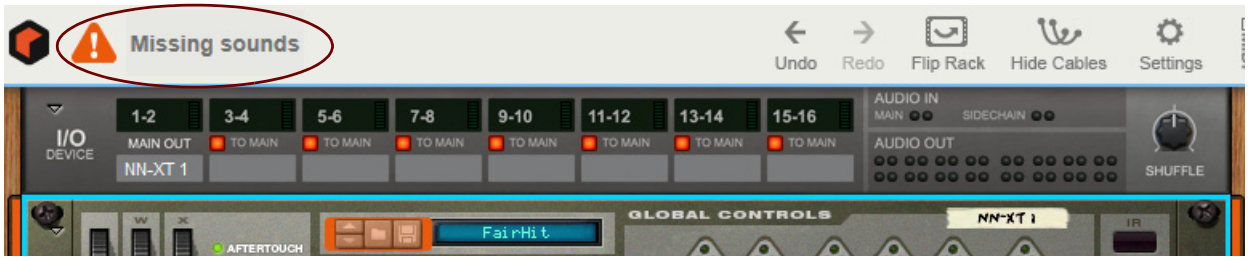


→ **To go back to the search result list, click the Back button:**



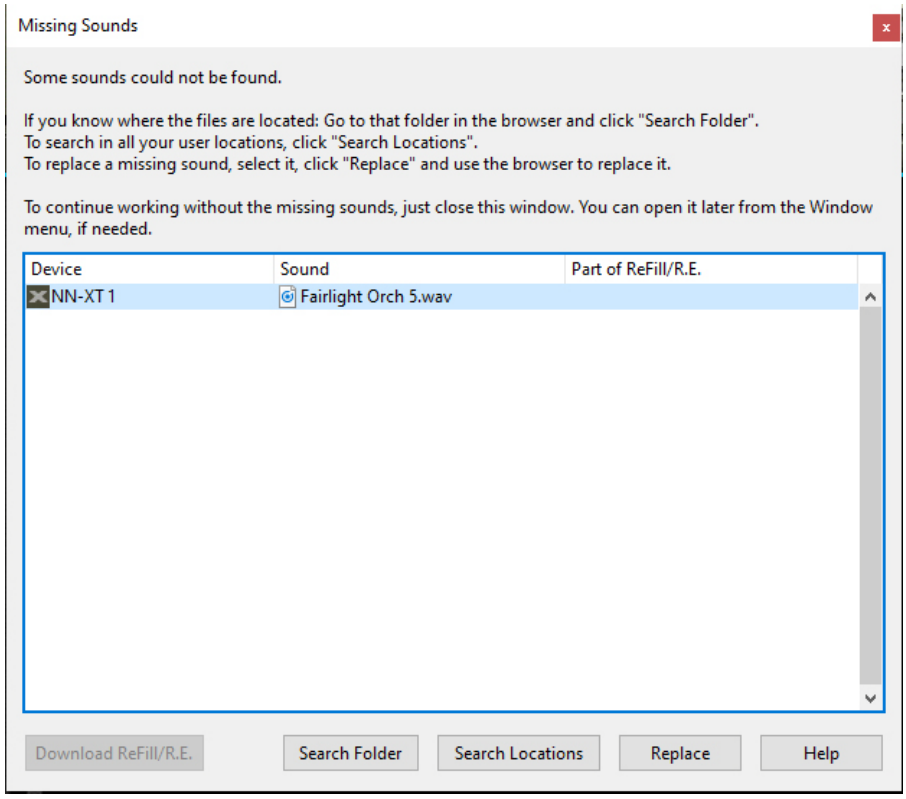
Handling Missing Sounds

If you load a patch for a sample-based instrument, and not all samples can be found, the Global Panel will show a Missing Sounds warning:



This may be because the samples have been moved or their folder renamed, etc.

1. **Click the Missing Sounds warning to open the Missing Sounds dialog:**



This lists all samples and loops that couldn't be found.

2. **Do one of the following:**

- **If you know where the missing sounds are, go to that folder in the Browser and then click "Search Folder".**
- **To search all your Browser locations for the missing sounds, click "Search Locations".**
- **To replace a single sample, select it in the list and click "Replace". Then find the replacement sample in the Browser and double click it.**

3. **When you're done, close the Missing Sounds dialog.**

Missing sounds that haven't been replaced will be indicated with an asterisk (*) before the file names in the sample browsers on the device panels.



About ReFills

A ReFill is a kind of component package for Reason, which can contain sounds and effects patches, samples, REX files, SoundFonts and demo songs. If you like, you can compare ReFills to ROM cards for a hardware synthesizer. On your computer, ReFills appear as large files with the extension ".rfl".

All sounds included with Reason Rack Plugin are embedded in three ReFills named "Reason Sounds", "Orkester Sounds" and "Factory Sounds", which were (downloaded and) copied to the hard disk during installation.

Additional ReFills are available for purchase. You can also download ReFills from other Reason users on the Internet, purchase them from other sample manufacturers, etc.

- **Samples (Wave and AIFF files) are compressed to about half their original file size when stored in ReFills, without loss of quality.**

In Reason Rack Plugin you can use the browser to list and access the embedded sounds and other components within the ReFills, just as if the ReFills were folders on your hard disk.

Furthermore, if a song makes use of components from ReFills, Reason Rack Plugin will tell you which ReFills are required.

Chapter 9
The I/O device

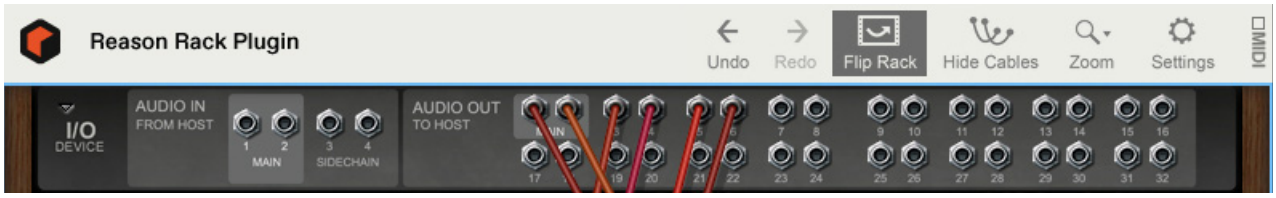


Introduction



At the top of the rack is the I/O device (for "input/output"). This handles the audio communication between the devices in the rack and the DAW host. It also includes a basic summing mixer for up to 8 stereo output channels and a control for setting the Shuffle amount used by some pattern devices.

The back panel



This is where the rack devices are connected.

The input jacks deliver audio from the DAW host to devices in the rack. This is most often the case when Reason Rack Plugin is used as an effect, and typically only the main (1-2) input jacks are used.

The output jacks deliver audio from devices in the rack to the DAW host. These are used both when Reason Rack Plugin is an instrument and an effect.

- **Devices you add will automatically be routed to free jacks on the I/O device.**

The front panel



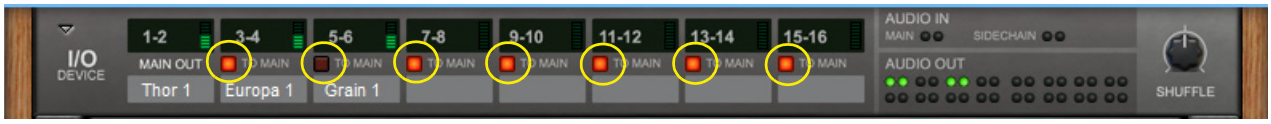
The first eight stereo outputs (output 1-16) have special settings on the front:

Level meters



These show the level of the signal received at the corresponding Output jack on the back panel.

Sum To Main (3-16 only)



When Sum to Main is on for an output pair, its signal is directed to the Main Out (1-2) instead, and summed with any other signals there.

- **If you want to layer several instrument devices on a single stereo channel in your DAW, leave "sum To Main" on.**
- **If you want different devices in your rack to be routed to different channels in your DAW's mixer, turn off "sum To Main" for these outputs.**

Name labels



These show the names of the connected devices. By default, devices have the name of the loaded patch or the device type, but you can change this by double-clicking the tape labels on the device panels and typing in a new name.

Audio In/Out indicators



These light up whenever audio signals are received from the DAW host (input) or sent back to the DAW host (output). Note that if you have connected an instrument to Output 3-4 and turned on sum To Main, the audio out indicators for 3-4 will not light up when you play the instrument! Instead, the audio out indicators for 1-2 will light up (because sum To Main sends the signal to the Main output 1-2).

The intensity of the lights indicate the audio signal levels.

Shuffle



Some devices features playback of patterns such as sequences and arpeggios. These can have a Shuffle mode, where 1/16th notes are shuffled for a swing feel. Examples of such devices include the Dual Arpeggio Player (see "Dual Arpeggio") and the Redrum drum computer (see "Redrum Drum Computer").

The amount of Shuffle for all such devices in a Reason Rack Plugin instance is set on the I/O device panel. Setting Shuffle to 50% results in a "straight" beat, with no swing applied. Setting the Shuffle to a value of 66% results in a perfect sixteenth-note triplet shuffle. Values between 50% and 66% have a less pronounced swing feel, and values greater than 66% are more exaggerated.

- **In the stand-alone version of Reason, this parameter is called "Global Shuffle" in the ReGroove Mixer.**
 If the reference manual for a device refers to Global Shuffle, it's the same as the Shuffle setting on the I/O device.



Chapter 10
Kong Drum Designer



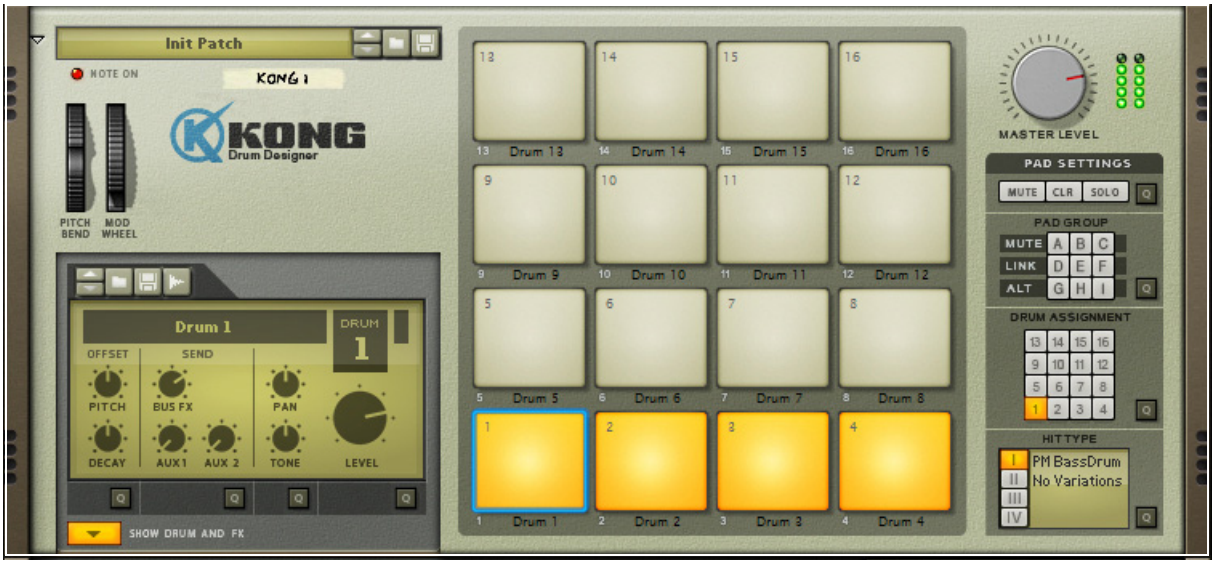
Introduction

The Kong Drum Designer gives the visual impression of a pattern-based drum machine, like the legendary MPC units. Indeed, it does have a matrix of 4 x 4 pads that are used for playing the sounds, just like the aforementioned classics. There are significant differences, however.

Kong features 16 drum "sound channels" that can host one drum sound each. Each drum sound can consist of a sound module routed through various types of FX and processing modules, allowing for completely open-ended sound design possibilities. Individual drum sounds can be saved as Drum Patches and complete drum kits can be saved as Kit Patches, allowing you to mix and match drum sounds and make up custom kits with ease.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

Overview



The Kong front panel.

Kong is an advanced drum sound synthesizer, sampler/sample player and REX loop/slice player with many unique features. The design could be described as semi-modular, in that the sound, FX and audio processing modules are open slots that allows you to select between an array of different sound generators, FXs and audio processor types.

As a result, Kong is capable of producing an astounding array of drum and percussion sounds - or any type of sound, for that matter. While it offers a lot of scope for serious sound design, it still has a straight-forward and user-friendly interface.

Kong also features audio inputs on the back panel. By connecting the output of another device to these inputs, you can use Kong's audio processing modules to process external sound. You can also route drum sounds for audio processing in external devices.

The Pad Section

The pad section features 16 pads. Each of these pads can be assigned to a separate Drum sound. You can also choose to assign several pads to one and the same Drum sound - or to link pads so that one pad will trigger several other pads as well. To the right of the pads is the Pad Settings area where you can control the pad assignments and behavior. See "Pad Settings".

The Drum Control Panel

The Drum Control Panel at the bottom left of the panel shows the name and "macro parameter" settings for the selected pad in the pad section. From the Drum Control Panel you can also load and save Drum Patches. See "The Drum Control Panel".

The Drum and FX Section

By clicking the Programmer button below at the bottom of the Drum Control Panel you can bring up the Drum and FX Section. Here is where you can edit your drum sounds and combine with various types of sound processors and FX. See "The Drum and FX section".

About using custom backdrops

As with the Combinator device, it is possible to customize the Kong front panel graphics with a user-designed skin. In the Reason Download section at the Reason Studios website is the "Combi and Kong Backdrop Templates" zip file, which can be used as starting point for designing your own Kong panel graphics. See the "Read Me.txt" file in the Backdrops folder for more details. Note that you should use separate backdrops for folded vs. unfolded states.

About file formats

Kong can read the following file types:

Kit Patches

A Kong Kit Patch (Windows extension ".kong") contains all settings for all 16 Drum sound channels, including file references to any used drum samples (but not the actual samples themselves). Switching patches is the same as selecting a new drum kit.

Drum Patches

A Kong Drum Patch (Windows extension ".drum") contains all settings for the selected Drum sound channel, including file references to any used drum samples (but not the actual samples themselves). Switching Drum Patches is the same as selecting a new drum sound.

Drum Samples

The audio file format support differs depending on which computer OS you are using.

The NN-Nano Sampler module in Kong can read and play back sample files of the following formats:

- **In Windows:**
 .wav, .aif, .mp3, .aac, .m4a and .wma.
- **In macOS:**
 .wav, .aiff, .3g2, .3gp, .mp1, .mp2, .mp3, .mpeg, .mpa, .snd, .au, .sd2, .ac3, .aac, .adts, .amr, .caf, .m4a .m4r and .mp4.
- **SoundFonts (.sf2)**
 SoundFonts are an open standard for wavetable synthesized audio, developed by E-mu systems and Creative Technologies.
- **REX file slices (.rx2, .rex, .rcy)**
 REX files are music loops created in the ReCycle program or when editing audio clips inline in Reason. The NN-Nano lets you load separate slices from REX files as individual samples.

- **Any sample rate and practically any bit depth.**

See "NN-Nano Sampler" for details.

REX Files

The Nurse Rex Loop Player module in Kong can read and play back files of the following formats:

- **REX files (.rx2, .rex, .rcy)**
 REX files are music loops created in the ReCycle program or when editing audio clips inline in Reason.

See "Nurse Rex Loop Player" for details.

Using patches

When you create a new Kong device it is loaded with a default kit. If you like you can use the default kit - or you can load another Kong Kit patch (or create one from scratch, by loading individual Drum patches). A Kong Kit patch contains settings for the 16 Drum channels, complete with parameter settings and file references to any samples used.

Loading a Kit Patch

To load a patch, use one of the following methods:

- **Use the browser to locate and open the desired patch.**
 To open the browser, select "Browse Kong Patches" from the Edit menu or device context menu, or click the folder button in the patch section on the device panel.



- **Once you have selected a patch, you can step between all the patches in the same folder by using the arrow buttons next to the patch name display.**
- **If you click and hold on the patch name display on the device panel, a pop-up menu will appear, listing all Kong Kit patches in all currently expanded folders in the Patch Browser.**
 This allows you to quickly select another patch without having to step through each one in turn.
- **Use the drag and drop method to drag Kong Kit Patch files from the Browser and drop on the Kong panel.**
 The Kong panel is dimmed in orange and a Patch Replace symbol appears in the center.

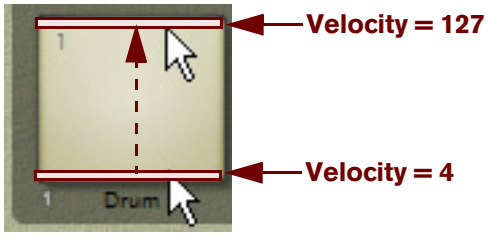
Checking the sounds in a Kit Patch

There are three ways you can listen to the sounds in a patch without using the main sequencer:

→ **By clicking the Pad buttons on the front panel.**

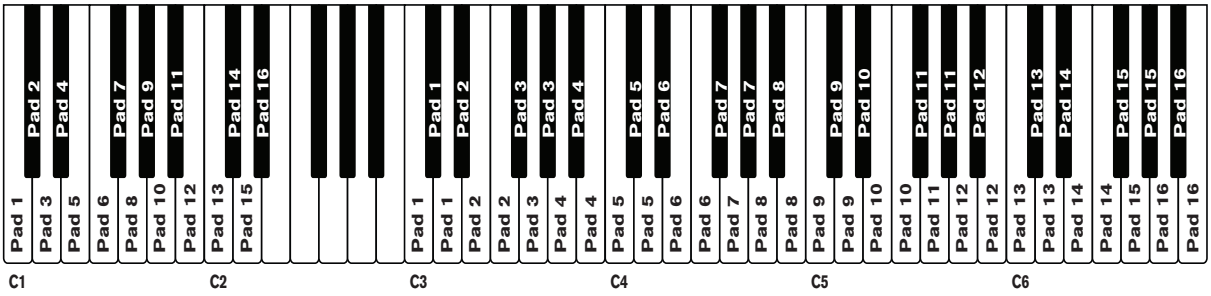


! **Note that the vertical click position on the pad determines the Velocity value. If you click towards the bottom of a pad, the velocity is low and at the top of each pad the velocity value is high.**



This will give you a good idea about the dynamics behavior of each drum sound. This also allows you to record in the main sequencer using the full dynamic range of each drum sound, even without a connected MIDI keyboard/ control surface.

→ **By playing the keys C1 to D#2 or C3 to B6 on your MIDI keyboard or on the On-screen Piano Keyboard.**



In the C1-D#2 range, each MIDI note will trigger one pad each, from Pad 1 to Pad 16. In the C3-B6 MIDI note range each pad can be triggered from three adjacent keys on your MIDI keyboard. C3-D3 trigs Pad 1, D#3-F3 trigs Pad 2 and so on. The C3-B6 note range is perfect if you want to play fast passages by triggering the same pad from several keys on your MIDI keyboard.

Creating a new Kit Patch

To create a patch of your own (or modify an existing patch), use the following basic steps:

1. **Click on the pad for the drum sound you want to load or replace.**
 A blue frame surrounds the selected pad.



2. **Click the folder button on the Drum Control Panel.**



→ **Alternatively, right-click (Win) or [Ctrl]-click (Mac) on the Pad and select "Browse Drum Patches..." from the context menu.**
The Patch Browser opens.

3. **Locate and open a Kong Drum Patch (extension '.drum') or a sample or REX file.**
You will find a selection of Kong Drum Patches in the Factory Sound Bank (in the Kong Drum Patches folder).

Loading a sample will automatically open it in an NN-Nano Sampler module (see "NN-Nano Sampler") and loading a REX file will automatically open it in a Nurse Rex Loop Player module (see "Nurse Rex Loop Player").

→ **Alternatively, drag a Kong Drum Patch file, a REX file, a sample or a REX slice from the Browser and drop on the Drum Control Panel - or on any desired drum pad.**
Depending on if you drop a Drum Patch file, a REX file or a sample/REX slice, the Drum Control Panel or pad is dimmed in orange or blue and a Patch/Sample Replace symbol appears in the center.

! **It is important that you drop REX files either on the Drum Control Panel or on a pad. Dropping it elsewhere will replace the entire Kong device with a Dr. Octo Rex device and load the REX file in this device instead!**

4. **Change some parameter settings for the drum sound channel using the knobs on the Drum Control Panel.**
These parameters are described in "The Drum Control Panel". Note that the Drum Control Panel parameters are "global" for each Drum channel. Each drum sound can consist of a number of different sound and FX modules, each with their separate set of parameters. Refer to "The Drum modules", "The Support Generator modules" and "The FX modules" for details about all the modules that can be used to build up a complete Drum sound.

5. **Repeat steps 1 and 4 for the other drum sound channels.**

6. **When you're satisfied with the drum kit, you can save the patch by clicking the Floppy Disk button in the patch section on the Drum panel.**
Note however, that you don't necessarily need to save the Drum patch - all settings are included when you save a Kong Kit Patch (see "Saving Kit Patches") and/or your song.

Creating an empty Kit Patch

To "initialize" the settings in the Kong, select "Reset Device" from the Edit menu or the device context menu. This removes all samples for all drum sound channels, and sets all parameters to their default values.

Saving Kit Patches

Saving patches is done in the same way as with any other Reason device - see "Loading patches" and "Saving patches".

! **Note that you don't have to save any of the 16 individual Drum patches first if you don't want to - all settings for each individual Drum patch are included in the Kong Kit patch.**

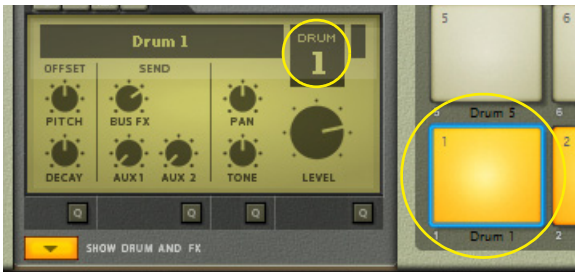
Pad Settings

In the Pad Settings section to the right of the Pad section you can perform various assignments and tricks pertaining to how the Drum channels should be controlled from the Pads.

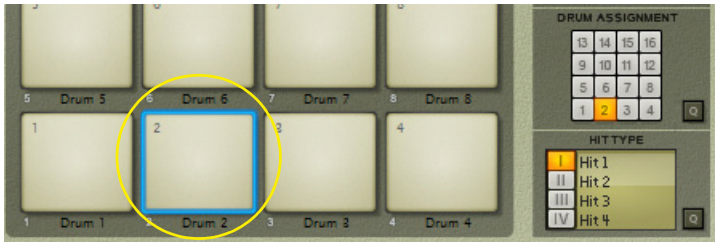
Assigning Drums to Pads

Kong features 16 pads and 16 Drum channels, as described earlier. Each pad can control a separate Drum sound channel. You can also assign several pads to control a single Drum sound channel. This is especially useful if you want to apply different settings, like Hit Types (see "Assigning Hit Type to Pads"), for each pad so that the controlled Drum sound channel responds differently. By default the 16 pads are assigned to their corresponding Drum sound channel; Pad 1 to Drum 1 and so on. If you want to change this assignment, proceed as follows:

1. **Select the desired Drum sound channel by clicking on its corresponding pad - or on the Pad name below the pad if you don't want the pad to trigger the sound.**
 A blue frame surrounds the selected pad and the corresponding Drum is displayed in the Drum section to the left.



2. **Select the other pad you want to control Drum 1 from.**
 In this example, we select Pad 2.



3. **Click on button 1 in the Drum Assignment section to assign Pad 2 to Drum 1.**

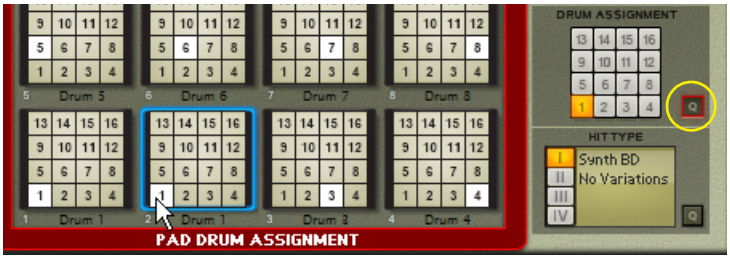


 Now, Pad 2 is also assigned to play Drum 1. Below Pad 2 it now says "Drum 1" to indicate the current assignment.

Assigning Drums to Pads using the Quick Edit function

If you want to assign several Drums to several pads quickly, you can do this by using the Quick Edit function.

1. **Click the Quick Edit button in the Drum Assign section.**



 Each Pad now shows the current Drum assignment.

2. **Change the Drum assignment clicking on the desired Drum channel number on each Pad.**
3. **When you are done, click the Quick Edit button or press [Esc] to exit to normal mode.**

Renaming Pads

→ **Double click on the Pad name below the corresponding Pad, enter a new name and press [Enter].**

Copying & Pasting Drums between Pads

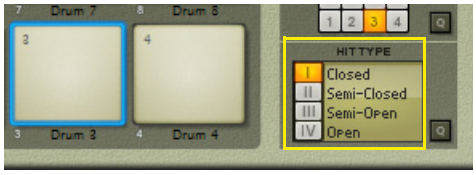
It's possible to copy a Pad with an assigned Drum and paste into another Pad location:

1. **Select the source Pad.**
2. **Select "Copy Drum Patch" from the context menu.**
3. **Select the destination Pad.**
4. **Select "Paste Drum Patch" from the context menu.**

Now, the complete Drum patch has been duplicated to the destination Pad and you can begin editing it as a separate patch.

Assigning Hit Type to Pads

If you have assigned several pads to the same Drum sound channel, you can choose a different Hit Type for each of the pads (where applicable). Depending on Drum sound type, some of the sounds can have up to four pre-defined Hit Types. These Hit Types are shown in the Hit Type display.



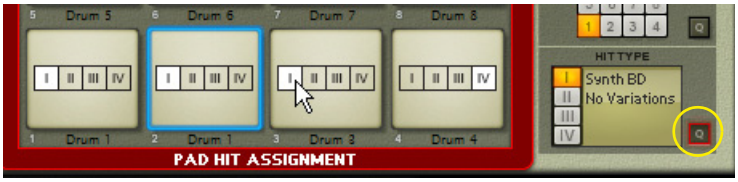
For example, a Synth Hi-Hat Drum sound has four Hit Types by default: "Closed", "Semi-Closed", "Semi-Open" and "Open". By selecting a different Hit Type for each of the pads assigned to the same Drum, you can create a very nice and "live" sound.

→ **To assign a Hit Type to a pad, select the pad and then select Hit Type by clicking the Hit Type button (or on the name in the display).**
 The Hit Type assignment is saved when you save your Kong Kit Patch and/or song.

Assigning Hit Type to Pads using the Quick Edit function

A quicker way of assigning Hit Type to several pads is by using the Quick Edit function.

1. **Click the Quick Edit button in the Hit Type section.**



Each Pad now shows the current Hit Type assignment.

2. **Change the Hit Type assignment clicking on the desired Hit Type number on each Pad.**
3. **When you are done, click the Quick Edit button or press [Esc] to exit.**

Muting and Soloing Pads



- **Click the Mute button to mute the assigned Drum for the selected Pad.**
 This will also mute MIDI control of the assigned Drum. Muted pads are displayed in red color.
- **Click the Solo button to solo the assigned Drum for the selected Pad.**
 Soloed pads are displayed in green color. All other pads are automatically muted. This also affects MIDI control of the Drum channels.
- **Click the CLR button to remove all Mute and Solo assignments.**

Muting and Soloing Pads using the Quick Edit function

A quicker way muting and soloing several pads is by using the Quick Edit function.

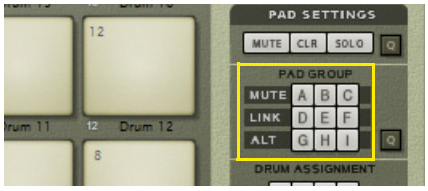
1. **Click the Quick Edit button at the top of the Pad Settings section.**



Each Pad will now show a Mute and Solo button.

2. **Click the Mute and/or Solo buttons on the desired Pads.**
3. **When you are done, click the Quick Edit button to exit.**

Working with Pad Groups



Kong features 9 Pad Groups, divided into 3 Mute Groups, 3 Link Groups and 3 Alt Groups. Each Pad can be assigned to one or more of these 9 Pad Groups independently. Pad Groups are useful if you, for example, want to trigger several pads from a single pad, have one pad mute another, or randomly trigger other pads from one pad.

Mute Groups

Mute Groups can be used if you want one pad to automatically mute another sound in the same Mute Group. For example, if you assign an open hi-hat and a closed hi-hat sound to the same Mute Group, playing on one pad will automatically mute the sound assigned to the other pad.

Link Groups

Pads assigned to the same Link Group will play together when you trigger any of the pads in that group.

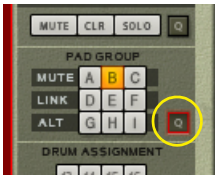
Alt Groups

If you play pads assigned to the same Alt Group, the pads will be triggered in a random fashion, one by one. It doesn't matter which pad you play in the group, the pad triggering is always random.

Assigning Pads to Pads Groups using the Quick Edit function
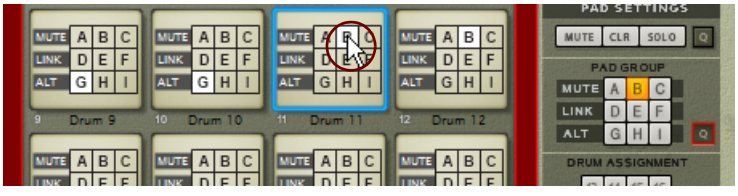
A quicker way of assigning several pads to Pad Groups is by using the Quick Edit function.

1. **Click the Quick Edit button in the Pad Group section.**



 Each Pad now shows the current Pad Group assignment.

2. **Edit the Pad Group assignment by clicking on the desired Pad Group letter on each Pad.**



 In the picture above, Pads 9 and 10 are assigned to Alt Group "G", which means they will trigger alternating when you play any of these Pads.

 Pads 11 and 12 are assigned to Mute Group "B", which means that playing Pad 11 will mute Pad 12 and vice versa.

3. **When you are done, click the Quick Edit button or press [Esc] to exit.**

The Drum and FX section



The Drum and FX section in Kong is built up of the Drum Control Panel and the Drum and FX section.

→ **Click the Show Drum and FX button below the Drum Control Panel to unfold the Drum and FX section.**
The Drum and FX section consists of five slots:

- **The Drum Module Slot.**
- **The FX1 Slot.**
- **The FX2 Slot.**
- **The Bus FX Slot.**
- **The Master FX Slot.**

The Drum, FX1 and FX2 slots are unique to each of the 16 Drum channels in Kong. The Bus FX and Master FX slots are shared between all Drum channels in the Kong device. You can activate/deactivate any of the slots by clicking the On button at the upper left of each slot.

Signal flow

The output signal from a Drum module is sent via the FX1 and FX2 Slots to the Bus FX, Master FX or to a pair of the individual outputs on the back of the Kong panel. There is also an internal Bus FX Send that can be used to send an audio signal from the Drum via the FX1 and FX2 Slots to the Bus FX. The Bus FX Slot can the hold e.g. a reverb module which can be used a send effect for all the Drum channels. As an extra bonus, you can also hook up an external effect device between the Bus FX and Master FX Slots, see "Using external effects with Kong".

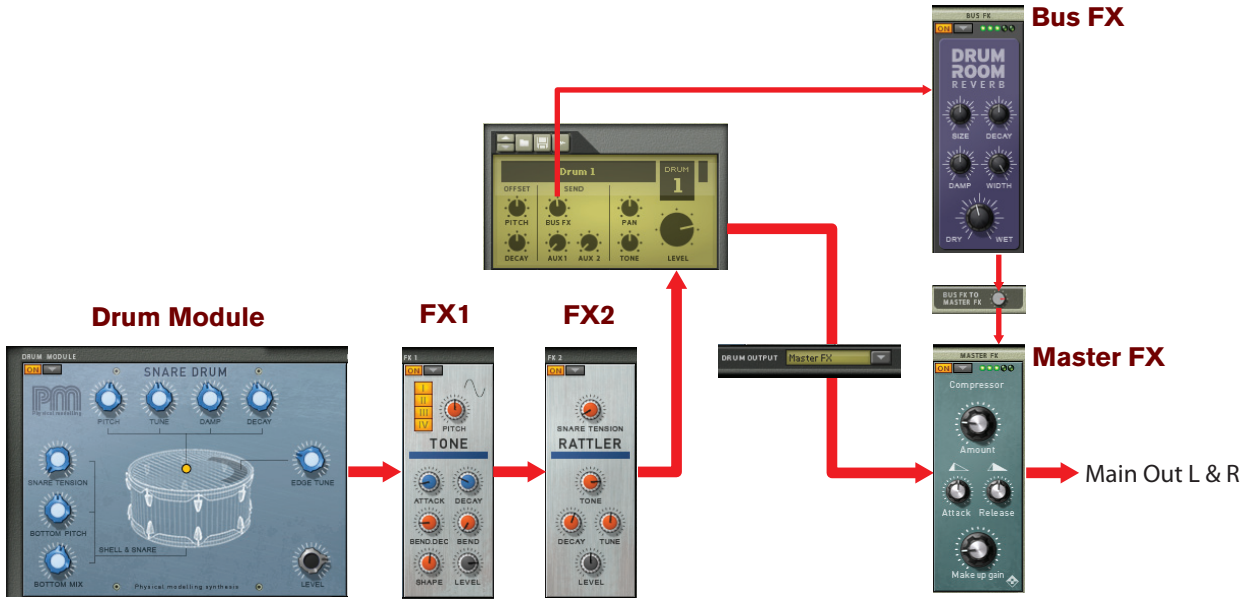
The signal routing in the Drum and FX section depends on the Drum Output selector setting at the bottom of the Drum and FX section:



The different signals flows are described in the following paragraphs:

Master FX Drum Output

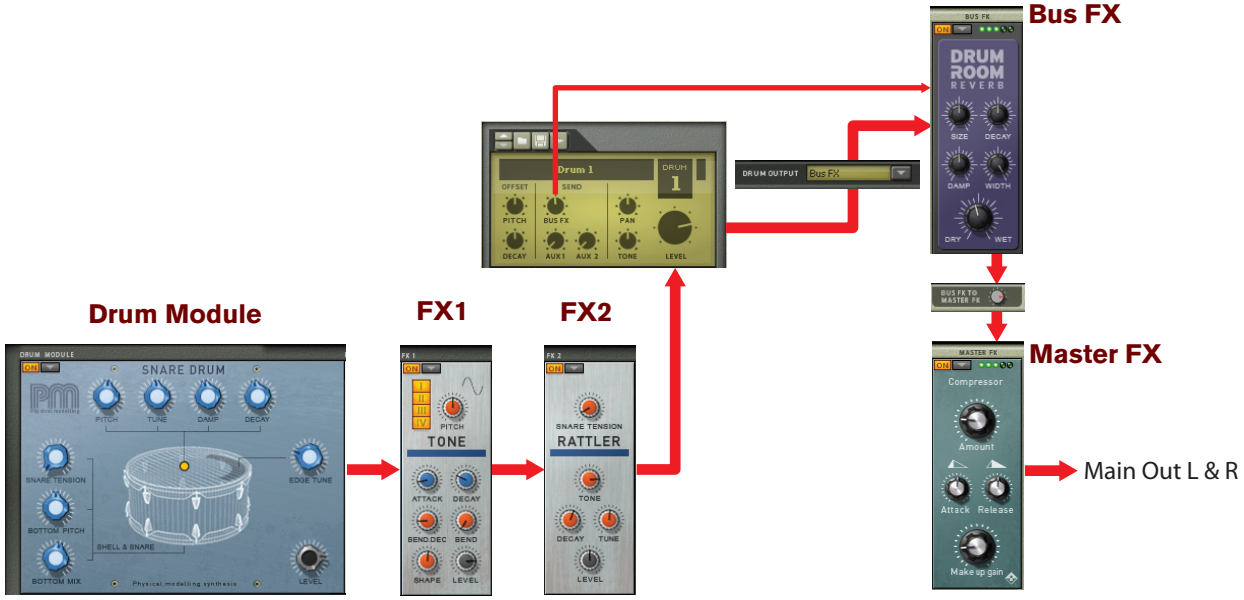
When the Drum Output is set to "Master FX", the signal flow is according to the picture below. If you are using a Bus FX, this is treated as a Send effect with the Bus FX level controlled by the Bus FX knob on the Drum Control Panel.



Signal flow when Drum Output is set to "Master FX".

Bus FX Drum Output

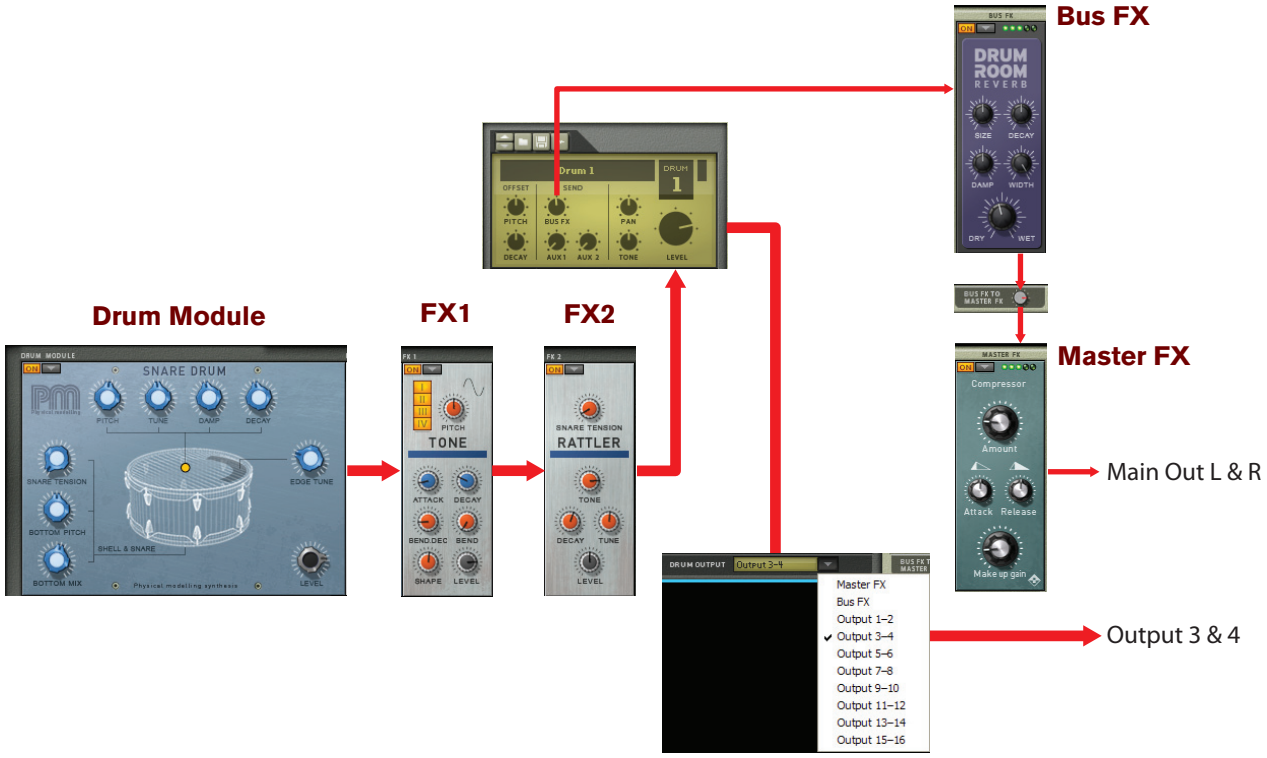
When the Drum Output is set to “Bus FX”, the signal flow is according to the picture below. Note that the Bus FX is now routed both as an Insert effect and as a Send effect at the same time. Therefore, it might be a good idea to set the Bus FX Send knob on the Drum Control Panel to zero in this configuration.



Signal flow when Drum Output is set to “Bus FX”.

Separate Out Drum Output

When the Drum Output is set to any of the separate output pairs “3-4” to “15-16”, the signal flow is according to the picture below. The signals to the selected separate output pair are taken directly after the FX2 via the Master Level knob. Note that the signal via the Bus FX and Master FX is still available on the Main Out L & R and can be controlled with the Bus FX Send knob on the Drum Control Panel.



Signal flow when Drum Output is set to any of the separate output pairs “3-4” to “15-16”.

The Drum Control Panel



The Drum Control Panel features a set of “macro controls” that affect parameters in each Drum. These controls scale the parameters in the Drum module and FX modules in the Drum and FX section. There are also some standard parameters that are identical for each Drum: Pan, Tone and Level.

- **The Pitch Offset knob affects the Pitch parameters in all Drum modules.**
 No FX modules are affected, even if they feature a Pitch parameter.

- **The Decay Offset knob affects the amplitude Decay or Release parameters in all Drum modules plus any FX modules that feature a Decay parameter.**
 For example, the reverb decay time in the Room Reverb FX module is affected by the Decay Offset parameter.
- **The Bus FX Send knob affects the signal level sent to the Bus FX Slot.**
 Depending on current Drum Output setting, the effect will be a little different - see the examples in "Signal flow".
- **The Aux 1 and Aux 2 Send knobs controls the level to any devices connected to the Aux 1 and Aux 2 Send Outputs on the back of the panel, see "Aux Send Out".**
 The signals to the Aux Send is tapped after the FX1 and FX2 Slots but before the Bus FX and Master FX Slots.
- **The Pan parameter controls the panning of the signal in the stereo panorama.**
 The Pan parameter affects the signal after the FX1 and FX2 Slots but before it is sent to the Bus FX and Master FX Slots.
- **The Tone parameter is a built-in filter (similar to the filter in Redrum).**
 The Tone parameter affects the signal after the FX1 and FX2 Slots but before it is sent to the Bus FX and Master FX Slots.

Editing the Drum Control Panel parameters using the Quick Edit function

A quicker way of editing the Drum Control Panel parameters for several Drum channels at once is by using the Quick Edit function. The Drum Control Panel features four Quick Edit buttons.

1. **Click the Quick Edit button below the Pitch and Decay Offset section.**



 Each Pad now shows the current Pitch and Decay Offset settings for each assigned Drum channel.

2. **Edit the Pitch and Decay Offsets by clicking and dragging the "crosshair" on the desired Pads.**



 The Decay Offset is on the horizontal X-axis and the Pitch Offset is on the vertical Y-axis, as shown in plain text on the big red frame around the Pad section. As you move the crosshair, the corresponding knobs on the Drum Control Panel move as well - and vice versa.

3. **When you are done, click the Quick Edit button or press [Esc] to exit - or click another Quick Edit button to change other sets of parameters.**

Loading and Saving Drum Patches



Loading and Saving Kong Drum patches (".drum") are done in the same way as with any other Reason device - see "Creating a new Kit Patch", "Loading patches" and "Saving patches".

A Kong Drum patch contains all parameter settings on the Drum Control Panel, including modules and parameter settings in the Drums and FX section - with references to any used samples.

It's also possible to load samples and REX loops in the Drum Control Panel section. Loading a sample will automatically open it in an NN-Nano Sampler module (see "NN-Nano Sampler") and loading a REX file will automatically open it in a Nurse Rex Loop Player module (see "Nurse Rex Loop Player").

The Drum Module slot



Each Drum channel in Kong has a main module slot - the Drum Module slot - to which you can load one of 9 different types of drum sound modules for designing drum sounds.

→ **Select Drum Module type by clicking the button to the right of the On button and selecting the module from the pop-up.**
the following Drum Module types can be selected: NN-Nano Sampler, Nurse Rex Loop Player, Physical Bass Drum, Physical Snare Drum, Physical Tom Tom, Synth Bass Drum, Synth Snare Drum, Synth Tom Tom and Synth Hi-Hat. See "The Drum modules" for details about each Drum module.

! **Note that only four pre-defined parameters per Drum Module can be automated!**

At the bottom below the Drum Slot is the Pitch Bend Range parameter which controls the Pitch Bend Range for the Drum Slot. This parameter is global for all types of Drum Modules but is unique to each of the 16 Drum channels.



The Pitch Bend Range knob for each of the 16 Drum channels

The FX slots



Each Drum channel also has 2 insert effect slots - the FX 1 and FX 2 Slots - to which you can load one of two different types of support sound generators or one of 9 different effect modules.

→ **Select Module type by clicking the button to the right of the On button and selecting module from the pop-up.** the following module types can be selected for the FX 1 and FX 2 Slots: Noise generator, Tone generator, Room Reverb, Transient Shaper, Compressor, Filter, Parametric EQ, Ring Modulator, Rattler, Tape Echo and Overdrive/ Resonator. See "The Support Generator modules" and "The FX modules" for details about each module type.

! **Note that only two pre-defined parameters per FX/Support Generator Module can be automated!**

• **For the Bus FX and Master FX slots, all module types except the Noise and Tone generators can be selected.**

The Drum modules

! **Note that only four pre-defined parameters per Drum Module can be automated!**

NN-Nano Sampler



The NN-Nano Sampler is based on the NN-XT Sampler and was designed to be ideal for drums and percussion sounds.

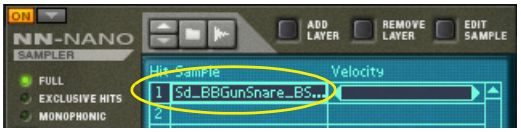
The NN-Nano can handle samples or sets of samples for each of the four different Hit Types described in "Assigning Hit Type to Pads". Each Hit Type can contain one or several samples which can be layered and/or altered and controlled individually via velocity.

Loading samples

1. **Select the Hit you want to load the sample(s) into by clicking in the display.**



2. **Click the Browse Samples (folder) button and select one or several WAV, AIFF or SoundFont Samples or REX slice files.**
3. **Click the Load button in the Browser.**
 The sample(s) are loaded in the selected Hit.

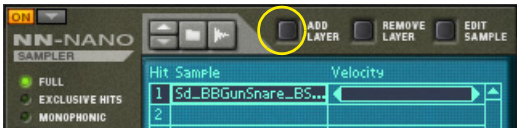


→ **Alternatively, drag a sample, a REX slice or a SoundFont file from the Browser and drop on the NN-Nano panel.**
The NN-Nano panel is dimmed in blue and a Sample Replace symbol appears in the center.

If you selected several samples in the Browser, these will be loaded as separate Layers in the selected Hit.

If you like you can load additional samples, either into another Hit or into a new Layer in the same Hit. To load a new sample in a new Layer in the same Hit, proceed as follows:

1. **Select the Hit and then click the Add Layer button.**



An additional space is created in the Hit.

2. **Select the empty Layer in the display and load a new sample according to the description in "Loading samples" above.**

The NN-Nano Sampler module in Kong can read and play back sample files of the following formats:

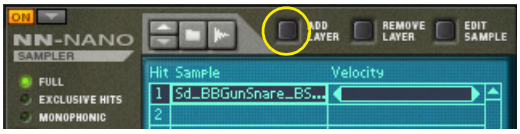
- **In Windows:**
 .wav, .aif, .mp3, .aac, .m4a and .wma.
- **In macOS:**
 .wav, .aiff, .3g2, .3gp, .mp1, .mp2, .mp3, .mpeg, .mpa, .snd, .au, .sd2, .ac3, .aac, .adts, .amr, .caf, .m4a .m4r and .mp4.
- **SoundFonts (.sf2)**
 SoundFonts are an open standard for wavetable synthesized audio, developed by E-mu systems and Creative Technologies.
- **REX file slices (.rx2, .rex, .rcy)**
 REX files are music loops created in the ReCycle program or when editing audio clips inline in Reason (see "Bounce Clip to REX Loop"). The NN-Nano lets load separate slices from REX files as individual samples.
- **Any sample rate and practically any bit depth.**

Replacing samples

→ **To replace one or several samples, select the sample(s) in the display and then load new samples according to the description in "Loading samples".**
This way it is possible to e.g. replace three selected samples with three new samples in one go.

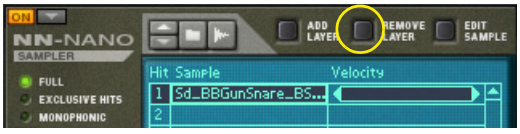
Adding and Removing Layers

→ **To add a new Layer to a Hit, select the Hit and click the Add Layer button.**



An additional space is created in the Hit for the created Layer.

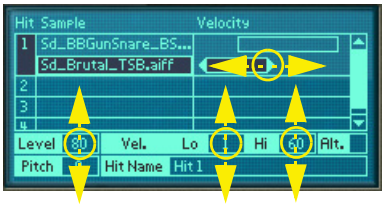
→ **To remove a Layer from a Hit, select the Layer and click the Remove Layer button.**



The Layer is removed together with its sample (if any).

Sample parameters

There are a number of parameters that are unique to each individual sample and Hit in the NN-Nano. These parameters are visible in the display for the selected (highlighted) sample:



- **Velocity**
 The Velocity range can be set, either by clicking and dragging the Velocity bar sideways to the right of the sample, or by clicking and dragging the Vel Lo and Hi values vertically at the bottom of the display.
- **Level**
 Set the sample level by clicking and dragging the Level value up or down in the display.
- **Pitch**
 Set the sample pitch by clicking and dragging the Pitch value up or down in the display.
- **Alt**
 Click the Alt box for several samples in the same Hit to make them play back alternating.
- **Hit Name**
 Edit the Hit Name if you like by clicking in the Hit Name box, typing in a new name and then pressing [Enter]. The name will appear in the Hit Type display on the main panel (see "Assigning Hit Type to Pads").

! **It's also possible to select multiple samples and edit them together. If the selected samples have different Level, Velocity, Range and/or Pitch values this is indicated by an "M" (for multiple) symbol next to the parameter:**



If you change the values of any of the "M" parameters, all selected samples will get the exact same value.

Global parameters

The parameters located on the panel, outside the display, are global and affect all samples in all Hit groups equally.



- **Polyphony**
 "Full" is, as the word implies, full polyphony. This means that all Hits can sound with full polyphony. Several Hits can also sound together if controlled from separate Pads that are assigned to different Hit Types.

 "Exclusive Hits" means that when one Hit plays it will automatically mute any other sounding Hits. The polyphony is still full within each Hit, though.

 "Monophonic" is... well, monophonic.
- **Mod Wheel**
 If you want the Mod Wheel to affect the pitch and/or decay of the sound, you can set this with the Mod Wheel -> Pitch and/or the Mod Wheel -> Decay knobs. Both parameters are bipolar (+/-).
- **Velocity**
 In the Velocity section you can control how the velocity should affect a number of parameters. The parameters are: Pitch, Decay, Level, Bend and Sample Start. All parameters are bipolar (+/-).
- **Pitch**
 Here you can set the global Pitch, Pitch Bend Amount and Pitch Bend Time for all samples. The Pitch and Pitch Bend Amount parameters are bipolar (+/-).
- **Osc**
 In the Osc section you can set the global Sample Start and Reverse parameters for all samples in the NN-Nano.
- **Amp Env**
 The Amp Env section contains an Attack-Decay Envelope and the global Level parameter for all samples. There is also an envelope trig mode selector for choosing between Gate and Trig mode. In Gate mode (the square symbol), the Decay time defines the minimum gate time. If you hold down a key or pad on your MIDI keyboard/control surface, the Decay stage will set in after you released the key/pad.

Nurse Rex Loop Player



The Nurse Rex Loop Player is based on the Dr. OctoRex Loop Player but has been modified to be ideal for playing and triggering drum and percussion sounds.

The Nurse Rex can load standard REX files and play back the loops and/or slices in a variety of ways depending on the selected Hit Type (see "Assigning Hit Type to Pads").

Loading REX files

1. **Click the Browse Samples (folder) button.**



2. **Select a REX file and click the Load button in the Browser.**

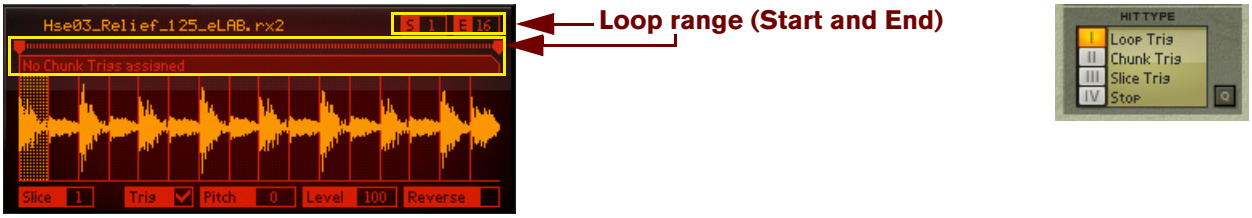
→ **Alternatively, drag a REX file from the Browser and drop on the Nurse Rex panel.**
The Nurse Rex panel is dimmed in orange and a Patch Replace symbol appears in the center.

The REX file is loaded in Nurse Rex with the loop shown in the display.

Hit Types (playback modes)

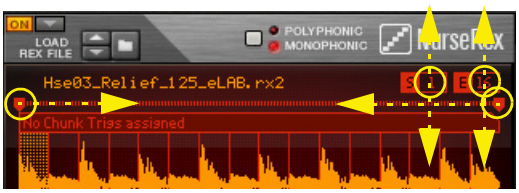
Depending on selected Hit Type, the REX loop will play back differently. The editing possibilities also differs depending on selected Hit Type for the assigned pad.

- **Loop Trig**
 In Loop Trig mode, you trigger the REX loop to play one single cycle every time you hit the assigned pad. Loop Trig can also be used together with the "Stop" mode on another pad to immediately stop the loop playback, see "Stop" below.



A REX loop with "Loop Trig" as Hit Type.

→ **Set Start and End slice, either by clicking and dragging the S and E numerical values up/down in the boxes, or by clicking and dragging the handles sideways in the "ruler" above the REX loop in the display.**



Different ways of editing the Start and End Slice values.

- **Chunk Trig**
 In Chunk Trig mode, you can assign several pads to play back shorter sections - chunks - of the REX loop. The number of chunks is determined by the number of pads you have assigned to the REX loop using the Chunk Trig Hit Type. The chunks are distributed in equal sections across the REX loop. Chunk Trig can also be used together with the "Stop" mode on another pad to immediately stop the chunk playback, see "Stop" below.

 In the picture below, we have assigned four pads to the same REX loop and we have selected "Chunk Trig" as Hit Type on all four pads:



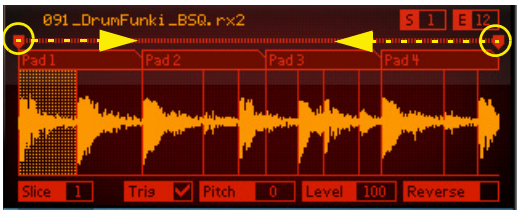
Four pads assigned to the same REX loop and Hit Type set to "Chunk Trig".

→ **Set the size of the chunks by clicking and dragging the right edges of the "tabs" above the REX loop in the display.**
Doing so will automatically move the start position of the subsequent chunk so that the chunks will always be adjacent to each other.



Editing the sizes of the chunks.

→ **Change the Start position of the first chunk and the End position of the last chunk by changing the REX loop Start and End values.**
In effect, this is the same as setting the overall REX loop start and end position.



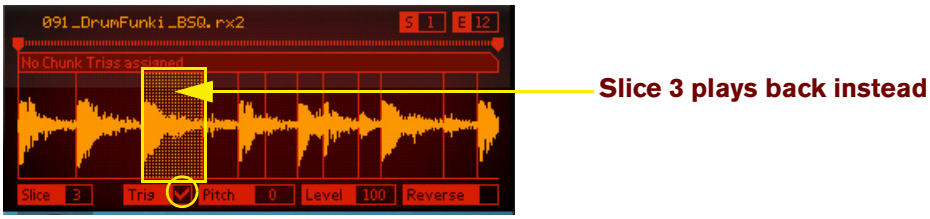
Editing the start position of the first chunk and end position of the last chunk.

- **Slice Trig**
In Slice Trig mode, you can assign a pad to play back one single slice of the REX loop - or several slices alternatingly. By default, Slice 1 of any REX loop loaded into the Nurse Rex is set to play back when you have selected "Slice Trig" as Hit Type.



A REX loop with "Slice Trig" as Hit Type.

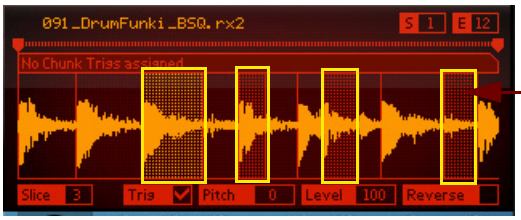
→ **Change the slice to play back by first removing the tick in the Trig checkbox and then clicking on another slice in the display and ticking the Trig checkbox for that slice.**



Slice 3 selected for playback in Slice Trig mode.

Another way of assigning a slice for playback, or to assign several slices to play back alternating, is by using the mouse in combination with the [Ctrl](Win)/[Cmd](Mac) key.

→ **Hold down [Ctrl](Win)/[Cmd](Mac) and click on the slice(s) in the display you want to assign or deassign.**



Slices 3, 5, 8 and 11 selected and will now play back alternating in Slice Trig mode

Slices 3, 5, 8 and 11 selected for playback in Slice Trig mode, forcing them to play back alternating.

Selected slices are displayed with a red background. The currently "focused" slice is displayed with an orange background. Selected slices also get their corresponding Trig checkbox ticked automatically.

- **Stop**
 The fourth Hit Type is named "Stop". The Stop mode can be used if you want to use a pad for immediately stopping the currently playing REX loop or Chunk. The Stop mode should be used in combination with any of the Hit Types "Loop Trig" or "Chunk Trig", otherwise it won't be useful.



"Stop" selected as Hit Type for a pad assigned to a Nurse Rex module.

1. **Assign one pad to a REX loop in Nurse Rex and select any of the Hit Types "Loop Trig" or "Chunk Trig".**
2. **Assign another pad to the same Nurse Rex module and select "Stop" as Hit Type.**
 Now, when you play the first pad, the loop or chunk will play. Once you hit the second pad, the loop/chunk playback will immediately stop.

Combining Hit Types

By combining the different Hit Types for the Nurse Rex module you can create really interesting setups. For example, you could load a REX loop and assign a couple of pads to the "Chunk Trig" Hit Type, one pad to Loop Trig, another one to Slice Trig and another pad to Stop. Playing the different pads can now generate really inspiring results. The picture below shows an example of this type of setup:

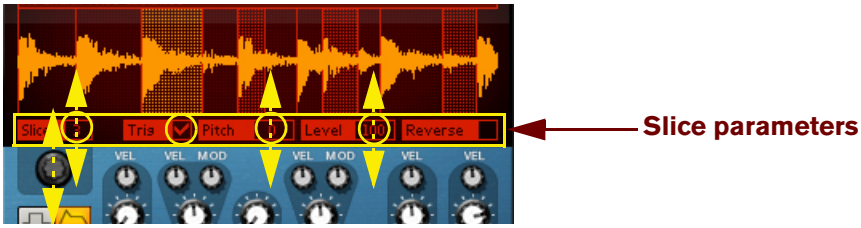


Eight pads assigned to the same Nurse Rex module, with the pads set to different Hit Types (in Quick Edit mode).

If we click the Hit Type Quick Edit button, we can see that Pad 1 is set to Loop Trig, Pads 2-6 are set to Chunk Trig, Pad 7 is set to Slice Trig and has four slices set to Trig in the REX loop display for alternate playback. Finally Pad 8 is set to Stop so we could stop the loop and chunks playback whenever we like.

Editing Slice Parameters

In the REX loop display you can edit parameters that are unique to each separate slice:



Slices parameters for a loaded REX loop.

→ **To select a slice for editing, click on the desired slice in the REX loop display.**
Alternatively, click and drag up/down in the Slice number box or use the Slice Select knob below the Slice item.

The Slice Parameters are:

- **Trig**
 Click the Trig check box for the slices you want to alternate between using the Slice Trig Hit Type.
- **Pitch**
 Set the pitch for each individual slice in the REX loop by clicking and dragging the Pitch value up/down.
- **Level**
 Set the level for each individual slice in the REX loop by clicking and dragging the Level value up/down.
- **Reverse**
 Click the Reverse box for the slices you want to play back backwards.

The Nurse Rex panel parameters

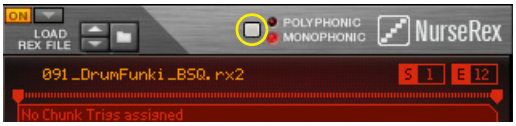
On the Nurse Rex panel you can edit parameters that are common to all slices in the loaded REX loop:



- **Env Type**
 Sets the amplitude envelope type to “Gate” or “ADSR” (Attack, Decay, Sustain, Release). In Gate mode, the gate time is set with the Decay parameter.
- **Attack with Velocity control**
 Sets the attack time for the amplitude envelope when ADSR is selected as Env Type. The attack time can also be velocity controlled according to the sensitivity set with the Vel knob.
- **Decay with Velocity and Modulation controls**
 Sets the decay time for the amplitude envelope when ADSR is selected as Env Type. When Gate is selected as Env Type, the Decay parameter sets the gate time. The decay/gate time can also be velocity controlled according to the sensitivity set with the Vel knob. You can also control the decay/gate time from the Mod Wheel with the amount set with the Mod knob.

- **Sustain**
 Sets the sustain level of the amplitude envelope when ADSR is selected as Env Type. In Gate mode, the Sustain parameter has no effect.
- **Release with Velocity and Modulation controls**
 Sets the release time for the amplitude envelope when ADSR is selected as Env Type. The release time can also be velocity controlled according to the sensitivity set with the Vel knob. You can also control the release time from the Mod Wheel with the amount set with the Mod knob. In Gate mode, the Release parameter has no effect.
- **Pitch with Velocity control**
 Sets the overall pitch of all slices in the REX loop. The pitch can be velocity controlled according to the Vel knob setting. A negative Vel setting will lower the pitch with increasing velocity and a positive setting will raise the pitch with increasing velocity.
- **Level with Velocity control**
 Sets the overall level of all slices in the REX loop. The level can be velocity controlled according to the Vel knob setting. A negative Vel setting will lower the level with increasing velocity and a positive setting will raise the level with increasing velocity.

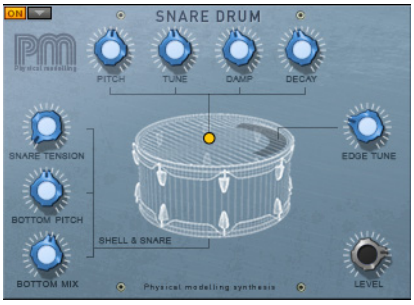
At the top of the Nurse Rex panel is a button for setting the polyphony:



- **Polyphonic means full polyphony**
 Retriggering the same slice/chunk will keep on adding more voices without muting any sustaining sounds.
- **Monophonic will make any new triggered loop/slice/chunk mute any currently playing/sustaining sound.**

Physical Bass Drum, Snare Drum and Tom Tom







The Physical Bass Drum, Snare Drum and Tom Tom use very faithful mathematical models for generating acoustic drum sounds. The sounds of the PM drums are generated using physical modelling; mathematical real-time calculations of physical acoustic phenomena. The physical modelling technique allows for a lot more creative freedom, and much wider sonic ranges, compared to sample playback.

General parameters

- **Level**
 This controls the overall output level of the Drum module to the FX1 and FX2 Slots (see "Signal flow"). The Level is also affected by velocity.

Drum head and shell parameters

The Physical Modelling Drums feature the following drum head and shell parameters:

- **Pitch**
 Sets the overall pitch of the drum. The Pitch parameter can be considered the total size of the drum and affects all other head and shell parameters.
- **Tune 1 and Tune 2 (PM Bass Drum and PM Tom Tom)**
 The Tune 1 and Tune 2 parameters set the drum's harmonic character, similar to the effect of individually adjusting the rim tension screws of the top drum head.
- **Tune (PM Snare Drum)**
 This controls the top drum head tension and thus affects the harmonic character of the sound.
- **Bend Amount (PM Bass Drum and PM Tom Tom)**
 This sets the dynamic "pitch bend" effect you get when hitting a drum head.
- **Damp**
 This controls the damping of the drum head.
- **Decay**
 The Decay parameter doesn't have any actual counterpart in real life. It simply controls the decay time of the drum sound.
- **Shell Level (PM Bass Drum and PM Tom Tom)**
 This controls how much of the drum shell sound should be present in the sound.
- **Shell Size (PM Tom Tom)**
 This controls the depth ("length") of the shell.
- **Edge Tune (PM Snare Drum)**
 This controls the head tuning when Hit Type 4 (Edge Hit) is selected for the pad (see "Assigning Hit Type to Pads").
- **Snare Tension (PM Snare Drum)**
 This controls the tension of the snare and the distance between the snare and bottom drum head.
- **Bottom Pitch (PM Snare Drum)**
 This controls the pitch of the bottom drum head.
- **Bottom Mix (PM Snare Drum)**
 This controls how much of the bottom head sound should be present in the drum sound.

Beater and Stick parameters

- **Density (PM Bass Drum)**
 This controls the "hardness" of the bass drum pedal beater.
- **Tone (PM Bass Drum and PM Tom Tom)**
 This is a filter which controls the tone of the hit.
- **Beater Level (PM Bass Drum)**
 This controls how much of the beater hit sound should be present in the drum sound.
- **Stick Level (PM Tom Tom)**
 This controls how much of the drum stick hit sound should be present in the drum sound.



Synth Bass Drum, Snare Drum and Tom Tom



The Synth Bass Drum, Snare Drum and Tom Tom use analog modelling to generate classic synth drum sounds. The Synth Tom Tom was faithfully modelled after a famous hexagonal shaped analog drum system from the 80's.

General parameters

- **Level**
 This controls the overall output level of the Drum module to the FX1 and FX2 Slots (see "Signal flow"). The Level is also affected by velocity.

Drum parameters

The Synth Drums feature the following parameters:

- **Pitch**
 This sets the overall pitch of the drum. The Noise pitch is not affected by this parameter.
- **Tone (Synth Bass Drum)**
 This is a filter similar to the one used in Redrum and affects the tone of the drum. The Noise is not affected by this parameter.
- **Attack (Synth Bass Drum)**
 Sets the attack time of the drum sound. This also affects the Noise.
- **Decay**
 Sets the Decay time of the drum sound. This also affects the Noise decay on the Synth Bass Drum and is added to the Noise Decay parameter on the Synth Snare and Synth Tom Tom drums. It is also added to the Harmonic Decay value on the Synth Snare Drum. The Decay time is also affected by velocity.
- **Harmonic Balance (Synth Snare Drum)**
 Sets the level balance between the fundamental tone and the harmonic tone.
- **Harmonic Frequency (Synth Snare Drum)**
 Sets the frequency of the harmonic tone.
- **Harmonic Decay (Synth Snare Drum)**
 Sets the decay time of the harmonic tone. This is also affected by the Decay parameter.
- **Click Frequency (Synth Bass Drum)**
 Sets the frequency of the click sound in the attack.
- **Click Resonance (Synth Bass Drum)**
 Sets the resonance amount of the click sound in the attack.
- **Click Level (Synth Bass Drum and Synth Tom Tom)**
 Sets the level of the click sound in the attack.

- **Bend Time (Synth Bass Drum and Synth Tom Tom)**
 Sets the time it should take to change the pitch from the Bend Amount value (se below) back to the original pitch.
- **Bend Amount (Synth Bass Drum and Synth Tom Tom)**
 Sets the upper pitch to bend from. The Bend Amount is also affected by velocity.
- **Noise Tone (Synth Snare Drum and Synth Tom Tom)**
 This is a filter which sets the frequency content of the noise.
- **Noise Decay (Synth Snare Drum and Synth Tom Tom)**
 This sets the decay of the noise in the sound. The Noise Decay is also affected by the regular Decay parameter.
- **Noise Mix (Synth Snare Drum and Synth Tom Tom)**
 Sets the noise level in the drum sound.

Synth Hi-hat



The Synth Hi-hat uses analog modelling to generate sounds. The Synth Hi-hat can be used for generating the typical hi-hat sounds of the early analog drum machines.

Parameters

- **Pitch**
 This sets the overall pitch of the hi-hat sound.
- **Decay**
 This sets the decay time of the hi-hat sound.
- **Level**
 This controls the overall output level of the Synth Hi-hat module to the FX1 and FX2 Slots (see “Signal flow”). The Level is also affected by velocity.
- **Click**
 This controls the click level in the attack of the hi-hat sound.
- **Tone**
 This is a filter similar to the one used in Redrum and affects the frequency content of the hi-hat sound.
- **Ring**
 Sets the level of the resonance peaks in the sound. The higher the value, the more “metallic” the sound.

The Support Generator modules

There are two types of Support Generator modules in Kong, one for generating noise and another one for generating a tone. The Support Generator modules can be used as companions to any of the Drum modules, or stand-alone. The Support Generators can be loaded into the FX1 and/or FX2 slots.

! **Note that only two pre-defined parameters per Support Generator module can be automated!**

Noise Generator



- **Hit Type buttons**
 These buttons allow you to choose for which Hit Type(s) the Noise generator should be active. By default, the Noise generator is active for all Hit Types (see "Assigning Hit Type to Pads").
- **Pitch**
 This sets the center pitch of the noise.
- **Attack**
 This sets the attack time of the noise.
- **Decay**
 This sets the decay time of the noise.
- **Reso**
 This sets the resonance amount of the noise around the center pitch.
- **Sweep**
 This sets the upper start pitch of the sweep range. The Sweep range is also affected by velocity.
- **Click**
 This sets the level of the click in the attack of the noise.
- **Level**
 This sets the overall level of the Noise generator. The level is also affected by velocity.

Tone Generator



- **Hit Type buttons**
 These buttons allow you to choose for which Hit Type(s) the Tone generator should be active. By default, the Tone generator is active for all Hit Types (see "Assigning Hit Type to Pads").
- **Pitch**
 This sets the pitch of the oscillator.
- **Attack**
 This sets the attack time of the tone.
- **Decay**
 This sets the decay time of the tone.
- **Bend Decay**
 This sets the decay time for the Bend.
- **Bend**
 This sets the upper start pitch of the bend range. The Bend range is also affected by velocity.
- **Shape**
 This sets the tonal character of the sound, from less to more harmonics.
- **Level**
 This sets the overall level of the Tone generator. The level is also affected by velocity.

The FX modules

The FX modules can be used in any of the FX1, FX2, Bus FX and Master FX slots.

! **Note that only two pre-defined parameters per FX Module can be automated!**

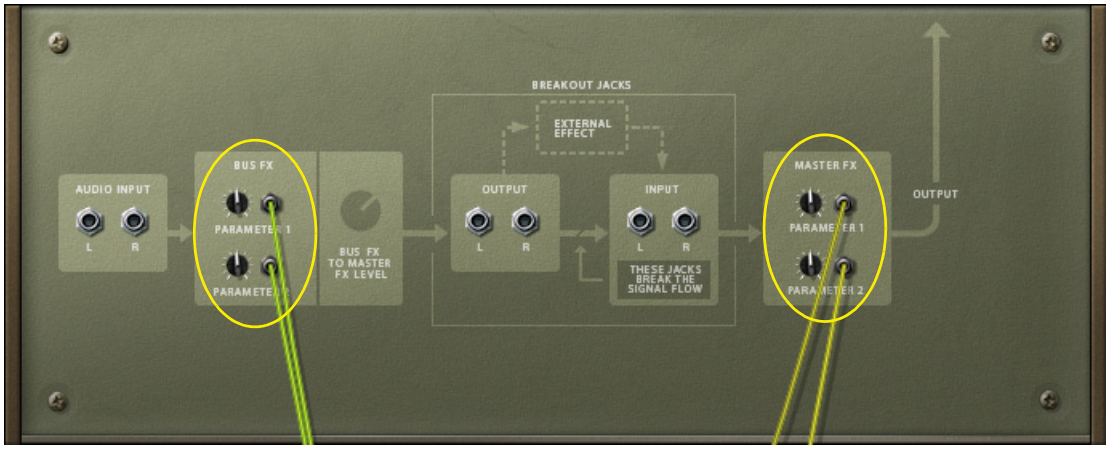
Using CV modulation of Bus FX and Master FX parameters

When the FX modules are used in the Bus FX and/or Master FX slots, it is possible to route external CV signals to the first two Effect module parameters for modulation. If you hover with the mouse over the first or second parameter of an FX module loaded in the Bus FX or Master FX slot, a tool tip appears:



The tool tip shows which CV modulation input on the back of the unfolded Kong panel will control that parameter. For FX modules loaded in the Bus FX slot, the tool tip displays “Bus FX P1: nn” for the first FX module parameter and “Bus FX P2: nn” for the second one. For FX modules loaded in the Master FX slot, the tool tip instead reads “Master FX P1: nn” for the first FX module parameter and “Master FX P2: nn” for the second one. The “nn” in the tool tip indicates the current parameter value.

By connecting cables to the CV modulation inputs on the back of the Kong panel, you can modulate the corresponding FX module parameters in the Bus FX and/or Master FX slots.



→ **Control the FX parameter modulation amounts with the corresponding attenuation knobs.**

If you decide to replace the FX modules in the Bus FX and/or Master FX slots, the modulation routing will be preserved - but the CV signals will now control the first two parameters of the replacement module(s).

Drum Room Reverb



The Drum Room Reverb is a reverb with a room-type reverb algorithm. It's perfect for adding ambience to single drum sounds or to the entire mix of all 16 drum channels. The parameters are as follows:

- **Size**
 This sets the "size" of the room, from small to large.
- **Decay**
 This sets the reverb decay time.
- **Damp**
 This sets the high frequency damping amount of the reverb effect, from none to heavy.
- **Width**
 This sets the stereo effect of the reverb, from mono to wide stereo.
- **Dry/Wet**
 This sets the mix between Dry (no effect) and Wet (reverb) signal.

Transient Shaper



The Transient Shaper is a type of dynamics processor which produces a result that could be compared to that of a compressor. As opposed to a "normal" compressor, the Transient Shaper mainly affects the signal's attack, or transients in the signal, making the signal transients cut through in the mix. The parameters are as follows:

- **Attack**
 A positive Attack value will produce an amplified attack/transient whereas a negative value will reduce the attack/transient volume.

- **Decay**
 This sets the decay time from amplification/attenuation back to normal amplitude level.
- **Amount**
 This controls the amplification amount. A high Amount in combination with a positive Attack value will produce a very pronounced attack/transient in the sound.

Compressor



The Compressor levels out the audio, by making loud sounds softer. To compensate for the volume loss, the Compressor has a make-up gain control for raising the overall level by a suitable amount. The result is that the audio levels become more even and the sounds can get more "power" and longer sustain. The parameters are as follows:

- **Amount**
 This sets the sensitivity of the compressor. A high amount will make the compressor more sensitive and react to weak input signals.
- **Attack**
 This sets how fast the compression should be applied to the incoming signal. A low value will make the compression set in immediately whereas a high value will let the attack/transients through before compression sets in.
- **Release**
 This sets how long it should take before the compressor lets the sound through unaffected again. Set this to short values for more intense, "pumping" compressor effects, or to longer values for a smoother change of the dynamics.
- **Make up gain**
 This sets the overall level compensation. A low value will produce a softer output signal whereas a high value will amplify the output signal.

Filter



The Filter is a state variable filter with a switch for selecting Lowpass, Bandpass or Highpass state. It has controls for cutoff/center frequency and resonance amount and can also be controlled from a built-in MIDI controlled envelope generator for sweeping the frequency. When used in the Bus FX Slot, MIDI Note E2 (#52) trigs the envelope. When used in the Master FX Slot, MIDI Note F2 (#53) trigs the envelope. The parameters are as follows:

- **Frequency**
 Sets the cutoff frequency in the LP and HP states and center frequency in the BP state.
- **Resonance**
 This sets the amplification amount of the frequencies around the cutoff/center frequency.
- **LP/BP/HP**
 Sets the state of the filter to either Lowpass, Bandpass or Highpass.
- **MIDI Trig EG Amount**
 This sets the amount of the MIDI controlled filter envelope. The Amount value is bipolar (+/-). Set to a positive value, the envelope will sweep the filter frequency from a high value down to the set Frequency value. Set to a negative Amount, the envelope will sweep the filter frequency from a low value up to the set Frequency value. The Amount is also affected by velocity.
- **MIDI Trig EG Decay**
 This sets the MIDI controlled envelope decay time.

Parametric EQ



The Parametric EQ is a single-band parametric equalizer with controls for center frequency, gain and bandwidth (Q-value). The parameters are as follows:

- **Frequency**
 Sets the center frequency of the equalizer.
- **Gain**
 Sets the amplification (positive Gain value) or attenuation (negative Gain value) around the center Frequency.
- **Q**
 Sets the bandwidth around the center Frequency, from wide to a narrow peak.

Ring Modulator



The Ring Modulator takes the input signal and multiplies it with an internal sinewave signal. The result is often a synthetic metallic sound. The Ring Modulator also features a MIDI controlled envelope generator for sweeping the internal sinewave frequency. When used in the Bus FX Slot, MIDI Note E2 (#52) trigs the envelope. When used in the Master FX Slot, MIDI Note F2 (#53) trigs the envelope. The parameters are as follows:

- **Frequency**
 Sets the frequency of the internal sinewave oscillator. The higher the frequency, the higher the resulting output signal pitch.
- **Amount**
 Sets the level of the internal sinewave oscillator. The higher the level, the more the ring modulation effect.
- **MIDI Trig EG Amount**
 This sets the amount of the MIDI controlled envelope. The Amount value is bipolar (+/-). Set to a positive value, the envelope will sweep the internal sinewave oscillator frequency from a high value down to the set Frequency value. Set to a negative Amount, the envelope will sweep the oscillator frequency from a low value up to the set Frequency value. The Amount is not affected by velocity.
- **MIDI Trig EG Decay**
 This sets the MIDI controlled envelope decay time.

Rattler



The Rattler adds the effect of a snare "attached" to whatever sound is fed through it. Using the Rattler in combination with other types of sounds than "usual" snare drum sounds can produce really interesting results! Ever played a snare bass drum, or a snare hi-hat, for example? The parameters are as follows:

- **Snare Tension**
 This sets the tension of the snare. Note that when the Snare Tension is increased, the effect will actually be less pronounced since the snare will have "less contact" with the sound source.
- **Tone**
 This is a filter similar to the one used in Redrum and affects the frequency content of the output signal.
- **Decay**
 This sets how long the snare will "ring".
- **Tune**
 This sets the snare tuning, from low to high, and affects the frequency content of the signal.
- **Level**
 This sets the overall level of the Rattler. The level is also affected by velocity.

Tape Echo



The Tape Echo is based on the principles of classic tape echo effects. The original tape echo effects were electro-mechanical devices that used an endless magnetic tape in combination with recording and playback heads inside the box. Depending on the speed of the tape, and on which playback heads were used, the echo repetition and echo patterns could be controlled. Later on, a lot of tape echo effects were replaced by digital delay effects. The Tape Echo in Kong simulates the classic tape echo effect and features the following parameters:

- **Time**
 This sets the time between the delays, from short to long.
- **Feedback**
 This sets the number of delay repetitions, from one to... many.
- **Wobble**
 This sets the tape speed wobbling effect. Since it emulates a magnetic tape, a wobbling speed also automatically produces a wobbling pitch of the signal.
- **Frequency**
 This sets the change in frequency of the delay repetitions. For every delay, the frequency content will shift according to the Frequency setting. A low value will make each repetition sound muddier than the previous one, whereas a high value will make each delay sound brighter.
- **Resonance**
 This sets the resonance amount of the delay repetitions. Depending on the Frequency parameter setting above, different frequencies will be amplified.
- **Dry/Wet**
 This is a traditional dry/wet parameter for controlling the relationship between unprocessed and processed signal.

Overdrive/Resonator



The Overdrive/Resonator is a combined distortion and resonator module. It can be used to add a nice distortion to the input signal. There is also a resonator section with a number of selectable characteristics, similar to the Body section in the Scream 4 Sound Destruction Unit. The parameters are as follows:

- **Drive**
 Sets the overdrive distortion amount.
- **Resonance**
 Sets the resonance amount for the resonator.
- **Size**
 Sets the size of the virtual “resonance chamber”, from small to large.
- **Model**
 Click to select one of five different resonator “body” characteristics.

Connections

On the back panel of Kong are a number of connectors. Many of these are CV/Gate related. Using CV/Gate is described in the chapter "Routing Audio and CV".



Sequencer Control

The Sequencer Control CV and Gate inputs allow you to play Kong from another CV/Gate device (typically a Matrix or a Redrum). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity.

Modulation Input

These control voltage (CV) inputs (with associated voltage trim pots), can modulate various Kong parameters from other devices. These inputs can control the following parameters:

- **Volume**
 This controls the Master Level in Kong.
- **Pitch**
 This controls the Pitch Bend wheel in Kong.
- **Mod**
 This controls the Mod Wheel in Kong.

Aux Send Out

The two stereo Aux Send Outputs can be used for connecting external effect devices for external signal processing. The signal levels to these Aux Send Outputs are controlled from the Aux 1 and Aux 2 Send knobs on the Drum Control Panel, see "The Drum Control Panel".

Gate In and Out

- **The Gate Inputs can receive a CV signal to trigger each of the 16 pads individually.**
 You are still able to control the pads from the panel and/or via MIDI even when the Gate Inputs are being used.
- **The Gate Outputs send out a CV Gate signal each time the corresponding pad is played.**
 The Gate signals can be used for triggering sounds in other devices.

Audio Out 3-16

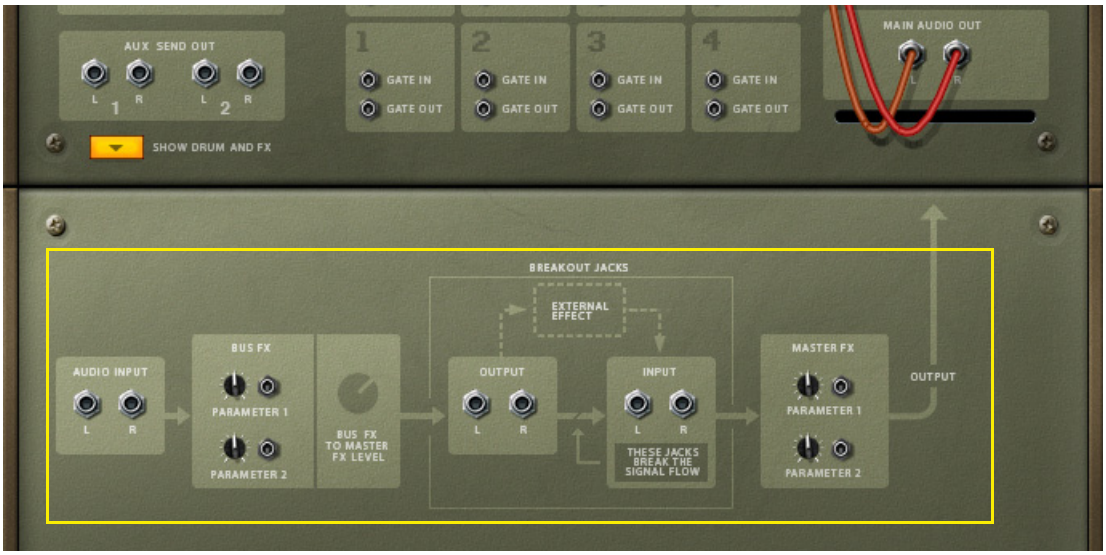
There are 14 separate audio output jacks on Kong's back panel - arranged as seven separate stereo pairs. These outputs are never auto-routed but can be manually connected and selected as individual outputs for any of the Drum channels by using the Drum Output selector in the Drum and FX section, see "Signal flow".

Main Audio Out

These are the main audio outputs. When you create a new Kong device, the Main Audio Output pairs auto-routed to the first available outputs of the I/O device.

Using Kong as an effect device

Besides using the wide array of internal sound possibilities in Kong, you can also use it as an external effect device. If you unfold the Drum and FX section and flip the rack around, a set of additional audio jacks are visible at the bottom of the back panel.



These audio jacks can be used for connecting external devices and processing their audio in Kong. As you can see, the signal flow for processing external audio is printed on the back panel. Even if you want to use Kong for processing external signals, you can still play and use its internal Drum channels just like before.

Proceed as follows to connect an external device for audio processing in Kong:

1. **Connect the outputs of your other device (a synth, for example) to the Audio Inputs to the left.**
 If your device only has a mono output, connect it to the Left Audio Input on Kong.

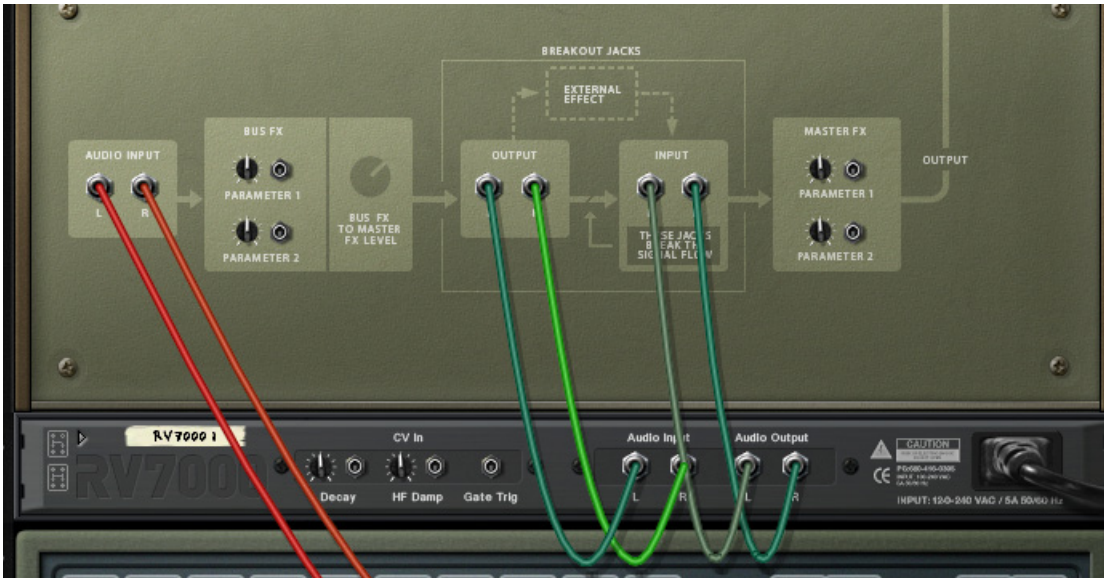
2. **Play a couple of notes on your other device.**
 The audio is now routed via Kong's Bus FX slot and further via the Master FX slot, to the Main Audio Outputs of Kong.
3. **Select suitable FX devices for the Bus FX and Master FX slots in Kong according to the descriptions in "The FX modules" and tweak the parameters to your liking.**

→ **By connecting CV signals to the Parameter inputs in the Bus FX and Master FX sections on the back of the panel, you can modulate the first two parameters on the selected FX modules, see "Using CV modulation of Bus FX and Master FX parameters".**
Attenuate the CV signal with the corresponding knobs next to the modulation jacks.

Using external effects with Kong

It's also possible to hook up an external effect device in the signal chain to process the Kong audio, or to process any external audio routed via the Audio Inputs to the left on the unfolded back panel of Kong.

→ **Connect the external effect device to the External Effect Outputs and Inputs.**
In the picture below, an RV7000 Reverb is connected to Kong's External Effect section allowing the RV7000 to process the signal between Kong's internal Bus FX and Master FX slots:



An RV7000 Reverb connected to Kong for processing the Kong audio signals

! **Note that if you have selected "Master FX" or "Separate Out" as output in the Drum Output selector, the BUS FX Send knob on the Drum Control Panel controls the signal level also to the External Effect, see "Signal flow".**

Chapter 11
Redrum Drum Computer



Introduction



At first glance, Redrum looks styled after pattern-based drum machines, like the legendary Roland 808/909 units. Indeed, it does have a row of 16 step buttons that are used for step programming patterns, just like the aforementioned classics. There are significant differences, however. Redrum features ten drum “channels” that can each be loaded with an audio file, allowing for completely open-ended sound possibilities. Don’t like the snare - just change it. Complete drum kits can be saved as Redrum Patches, allowing you to mix and match drum sounds and make up custom kits with ease.

About file formats

Redrum reads two basic types of files:

Redrum Patches

A Redrum patch (Windows extension ".drp") contains all settings for all ten drum sound channels, including file references to the used drum samples (but not the actual drum samples themselves). Switching patches is the same as selecting a new drum kit.

Drum Samples

The audio file format support differs depending on which computer OS you are using.

Redrum can read audio files in the following formats:

- **In Windows:**
 .wav, .aif, .mp3, .aac, .m4a and .wma.
- **In macOS:**
 .wav, .aiff, .3g2, .3gp, .mp1, .mp2, .mp3, .mpeg, .mpa, .snd, .au, .sd2, .ac3, .aac, .adts, .amr, .caf, .m4a .m4r and .mp4.
- **SoundFonts (.sf2)**
 SoundFonts are an open standard for wavetable synthesized audio, developed by E-mu systems and Creative Technologies.
- **REX file slices (.rx2, .rex, .rcy)**
 REX files are music loops created in the ReCycle program or when editing audio clips inline in Reason. Redrum lets you load separate slices from REX files as individual samples.
- **Any sample rate and practically any bit depth.**

Using patches

When you create a new Redrum device it is loaded with a default kit. If you like you can program a pattern and play back using the default kit - or you can load another Redrum patch (or create one from scratch, by loading individual drum samples). A Redrum patch contains settings for the ten drum sound channels, complete with file references to the drum samples used.

! **Redrum patterns are not part of the patch! If you want to save Redrum patches complete with patterns, create a Combinator containing the Redrum and save the Combi patch.**

Loading a patch

To load a patch, use one of the following methods:

→ **Use the Browser to locate and load the desired patch.**
To open the browser and set browse focus to the Redrum device, select "Browse Redrum Patches" from the device context menu, or click the folder button in the patch section on the device panel.



→ **Once you have selected a patch, you can step between all the patches in the same folder by using the arrow buttons next to the patch name display.**

→ **If you click on the patch name display on the device panel, a pop-up menu will appear, listing all patches in the current folder.**
This allows you to quickly select another patch in the same folder, without having to step through each one in turn.

→ **Drag a Redrum (.drp) patch from the Browser and drop on the device panel.**
The panel is dimmed in orange and a Patch Replace symbol appears in the center.

Checking the sounds in a patch

There are two ways you can listen to the sounds in a patch without programming a pattern:

→ **By clicking the Trigger (arrow) button at the top of each drum sound channel.**



→ **By playing the keys C1 to A1 on your MIDI keyboard.**
C1 plays drum sound channel 1 and so on. See also "Using Redrum as a sound module".

Both these methods play back the drum sample for the corresponding drum sound channel, with all settings for the sound applied.

Creating a new patch

To create a patch of your own (or modify an existing patch), you use the following basic steps:

1. **Click the folder button for a drum sound channel.**
 The Redrum sample browser opens.



2. **Locate and load a drum sample.**
 You will find a large number of drum samples in the Factory Sound Bank (in the folder Redrum Drum Kits/xclusive drums-sorted). You can also load other samples in any supported format.

→ **Alternatively, drag a sample file from the Browser and drop on the desired sound channel section.**
The sound channel is dimmed in blue and a Sample Replace symbol appears in the center.

3. **Make the desired settings for the drum sound channel.**
 The parameters are described in "Redrum parameters".
4. **Repeat steps 1 and 3 for the other drum sound channels.**
5. **When you're satisfied with the drum kit, you can save the patch by clicking the Floppy Disk button in the patch section on the device panel.**
 Note however, that you don't necessarily need to save the patch - all settings are included when you save the song.

Loading REX file slices

Loading slices from within a REX file is done much in the same way as loading "regular" samples:

1. **Open the Browser as described above.**
2. **Browse to a REX file.**
 Possible extensions are ".rx2", ".rex" and ".rcy".
3. **Unfold the REX file.**
 The browser will now display a list of all the separate slices within the REX file.
4. **Select the desired slice and click the Load button in the Browser.**
 The slice is loaded into the Redrum.

→ **Alternatively, drag a REX slice file from the Browser and drop on the desired sound channel section.**
The sound channel is dimmed in blue and a Sample Replace symbol appears in the center.

Creating an empty patch

To "initialize" the settings in the Redrum, select "Reset Device" from the device context menu. This removes all samples for all drum sound channels, and sets all parameters to their default values.

Programming patterns

Pattern basics

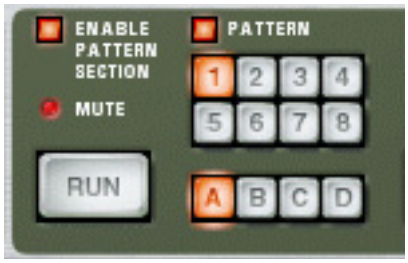
Redrum contains a built-in pattern sequencer. Unlike the main sequencer in Reason, the Redrum sequencer repeatedly plays back a pattern of a specified length. The typical analogy in the "real world" is a drum machine which plays drum patterns, usually one or two bars in length.

Having the same pattern repeat throughout a whole song may be fine in some cases, but most often you want some variations. The solution is to create several different patterns and program pattern changes (automatic switching from one pattern to another) at the desired positions in the song.

How the Redrum pattern sequencer integrates with the main sequencer

The built-in pattern sequencer in the Redrum interacts with the main sequencer in the following ways:

- **The tempo set on the transport panel is used for all playback.**
 If tempo automation is used in the main sequencer, Redrum will follow this.
- **If you start playback for the main sequencer (on the transport panel), Redrum will automatically start as well (provided the pattern sequencer hasn't been disabled - see below).**
- **You can mute and solo Redrum tracks in the sequencer.**
 If the Redrum has a track in the sequencer and you mute this track, Redrum will automatically be muted as well. This is indicated by a Mute indicator on the device panel.



This Redrum device is muted.

- **You can also run Redrum separately (without starting the main sequencer) by clicking the Run button on the device panel.**
 This starts the built-in pattern sequencer in the device. To stop playback, click the Run button again or click the Stop button on the Transport panel.



The Run button on the Redrum.

- **If you are running Redrum separately and start playback of the main sequencer, the pattern device will automatically restart in sync with the sequencer.**
- **Pattern changes can be controlled by pattern automation in the main sequencer.**

- **The sound sources can also be played by the main sequencer, or via MIDI.**
 You can combine the built-in pattern playback with playback from the main sequencer or via MIDI. For example, this allows you to add variations or fills to a basic pattern.
 It is also possible to disable the pattern sequencer totally, converting the device to a pure sound module. This is done by deactivating the Enable Pattern Section switch.



Selecting patterns

The Redrum has 32 pattern memories, divided into four banks (A, B, C, D).



The Bank and Pattern buttons for the Redrum pattern sequencer.

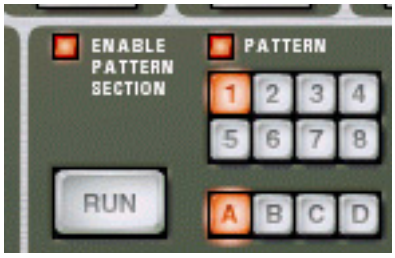
- **To select a pattern in the current bank, click on the desired Pattern button (1-8).**
 If you like, you can assign computer key commands and/or MIDI messages to pattern selection.
- **To select a pattern in another bank, first click the desired Bank button (A, B, C, D) and then click the Pattern button.**
 Nothing happens until you click the Pattern button.
- **If you select a new pattern during playback, the change will take effect on the next downbeat (according to the time signature set in the transport panel).**
 If you automate pattern changes in the main sequencer, you can make them happen at any position.
- **Note that you cannot load or save patterns - they are only stored as part of a song.**
 However, you can move patterns from one location to another (even between songs) by using the Cut, Copy and Paste Pattern commands on the context menu.

Pattern tutorial

If you are unfamiliar with step programming patterns, the basic principle is very intuitive and easy to learn. Proceed as follows:

1. **Load a Redrum patch, if one isn't already loaded.**
2. **Make sure an empty pattern is selected.**
 If you like, use the Clear Pattern command on the device context menu to be sure.

3. **Make sure that the “Enable Pattern Section” and the “Pattern” buttons are activated (lit).**



4. **Press the “Run” button.**
 There will be no sound, as no pattern steps have been recorded yet. But as you can see, the LEDs over the Step button light up consecutively, moving from left to right, and then starts over. Each Step button represents one “step” in the Pattern.
5. **Select a Redrum channel, by clicking the “Select” button at the bottom of the channel.**
 The button lights up, indicating that this channel and the drum sound it contains is selected.



6. **While in Run mode, press Step button 1, so that it lights up.**
 The selected sound will now play every time Step 1 is “passed over”.
7. **Clicking other Step buttons so they light up will play back the selected sound as the sequencer passes those steps.**
 Clicking on a selected (lit) step button a second time removes the sound from that step and the button goes dark again. You can click and drag to add or remove steps quickly.
8. **Select another Redrum channel to program steps for that sound.**
 Selecting a new sound or channel also removes the visual indications (static lit buttons) of step entries for the previously selected sound. The step buttons always show step entries for the currently selected sound.
9. **Continue switching between sounds, and programming steps to build your pattern.**
 Note that you can erase or add step entries even if Run mode isn't activated.

Setting pattern length

You may want to make settings for Pattern length, i.e the number of steps the pattern should play before repeating:



- **Use the “Steps” spin controls to set the number of steps you wish the pattern to play.**
 The range is 1 to 64. You can always extend the number of steps at a later stage, as this will merely add empty steps at the end of the original pattern. You could also make it shorter, but that would (obviously) mean that the steps “outside” the new length won't be heard. These steps aren't erased though; if you raise the Steps value again, the steps will be played back again.

About the “Edit Steps” Switch

If you set the pattern length to more than 16 steps, the pattern steps following after the 16th won't be visible, although they will play back. To view and be able to edit the next 16 steps, you have to set the Edit Steps switch to 17-32. To see and edit steps beyond 32 you set the switch to 33-48, and so on.

Setting pattern resolution



Redrum always follows the tempo setting on the transport panel, but you can also make Redrum play in different “resolutions” in relation to the tempo setting. Changing the Resolution setting changes the length of each step, and thereby the “speed” of the pattern.

Step dynamics

When you enter step notes for a drum sound, you can set the velocity value for each step to one of three values: Hard, Medium or Soft. This is done by setting the Dynamic switch before entering the note.



The color of the step buttons reflect the dynamics for each step. Soft notes are light yellow, Medium notes are orange and Hard are red.

- **When the Medium value is selected, you can enter Hard notes by holding down [Shift] and clicking.**
 In the same way, you can enter Soft notes by holding down [Option] (Mac) or [Alt] (Windows) and clicking. Note that this doesn't change the Dynamic setting on the device panel - it only affects the notes you enter.
- **When you use different dynamics, the resulting difference in the sound (loudness, pitch, etc.), is governed by the “VEL” knob settings for each drum channel (see “Redrum parameters”).**
 If no velocity amount is set for a drum channel, it will play back the same, regardless of the Dynamic setting.
- **To change the dynamics for an already programmed step, set the switch to the dynamic value you wish to change it to and click on the step.**

! **Note that if you are triggering Redrum via MIDI or from the main sequencer, the sounds will react to velocity like any other audio device. The Dynamic values are there to offer velocity control when using the built-in pattern sequencer.**

Pattern Shuffle

Shuffle is a rhythmic feature, that gives the music a more or less pronounced swing feel. It works by delaying all sixteenth notes that fall in between the eighth notes. You can activate or deactivate shuffle individually for each Redrum pattern by clicking the Shuffle button on the device panel.



The amount of shuffle is set globally with the Shuffle control on the I/O device - see "Shuffle".

Flam



A flam is when you double-strike a drum, to create a rhythmic or dynamic effect. Applying flam to a step entry will add a second "hit" to a drum sound. The flam amount knob determines the delay between the two hits.

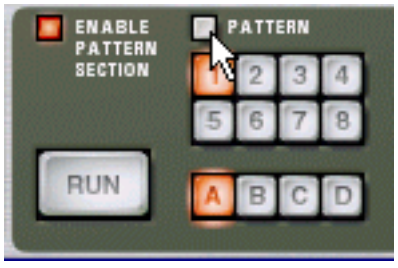
To add a flam drum note, proceed as follows:

1. **Activate flam by clicking the Flam button.**
2. **Click on a step to add a note (taking the Dynamic setting into account as usual).**
 A red LED is lit above the step to indicate that flam will be applied to that step.
3. **Use the Flam knob to set the desired amount of flam.**
 The flam amount is global for all patterns in the device.

→ **To add or remove flam to or from an existing step note, click directly on the corresponding flam LED.**
You can also click and drag on the LEDs to add or remove several flam steps quickly.

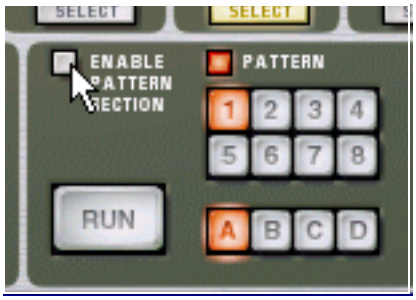
→ **Applying flam to several consecutive step entries is a quick way to produce drum rolls.**
By adjusting the Flam knob you can create 1/32 notes even if the step resolution is 1/16, for example.

The Pattern Enable switch



If you deactivate the "Pattern" button the pattern playback will be muted, starting at the next downbeat (exactly as if you had selected an empty (silent) pattern). For example, this can be used for bringing different pattern devices in and out of the mix during playback.

The Enable Pattern Section switch



If this is off, Redrum will function as a pure "sound module", i.e. the internal Pattern sequencer is disengaged. Use this mode if you wish to control Redrum exclusively from the main sequencer or via MIDI (see "Using Redrum as a sound module").

Pattern functions

When a Redrum device is selected, you will find some specific pattern functions on the Edit menu (and on the device context menu):

Function	Description
Shift Pattern Left/Right	These functions move all notes in the pattern one step to the left or right.
Shift Drum Left/Right	The Shift Drum functions move all notes for the selected drum channel (the channel for which the Select button is lit) one step to the left or right.
Randomize Pattern	Creates a random pattern. Random patterns can be great starting points and help you get new ideas.
Randomize Drum	Creates a random pattern for the selected drum sound only - the notes for the other drum sound channels are unaffected.
Alter Pattern	The Alter Pattern function modifies the selected pattern by "shuffling" the current pattern notes and redistributing them among the drum sounds at random. This creates a less chaotic pattern than the "Randomize Pattern" function. Note that there must be something in the pattern for the function to work on - using an Alter function on an empty pattern will not do anything.
Alter Drum	Works like the "Alter Pattern" function, but affects the selected drum sound only.

Chaining patterns

When you have created several patterns that belong together, you most probably want to make these play back in a certain order. This is done by recording pattern automation into the main sequencer.

Copy MIDI Notes to a sequencer track

It's possible to copy the MIDI Notes of the currently selected Pattern to a track in your DAW's sequencer:

1. **Click to select the desired Pattern.**
2. **Click the "Drag MIDI Notes to track in sequencer" icon and drag to the destination track in the sequencer:**



! **Be sure to disable the Enable Pattern Section function on the Redrum panel afterwards, to avoid "doubled notes" during playback.**

Redrum parameters

Drum sound settings

Redrum features ten drum sound channels that can each be loaded with a sample. Although they are basically similar, there are three "types" of drum sound channels, with slightly different features. This makes some channels more suitable for certain types of drum sounds, but you are of course free to configure your drum kits as you like.

On the following pages, all parameters will be listed. If a parameter is available for certain drum sound channels only, this will be stated.

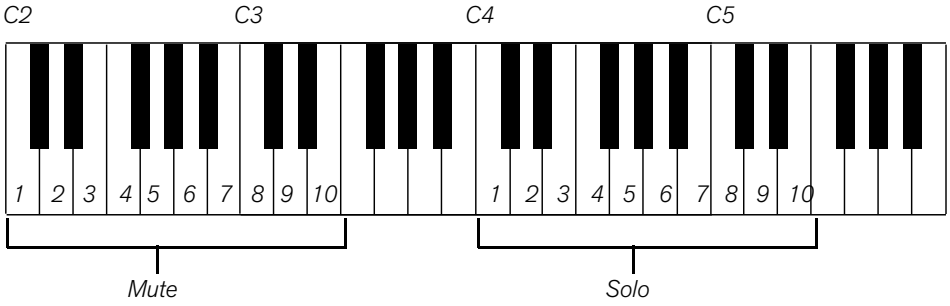
Mute & Solo



At the top of each drum sound channel, you will find a Mute (M) and a Solo (S) button. Muting a channel silences its output, while Soloing a channel mutes all other channels. Several channels can be muted or soloed at the same time.

You can also use keys on your MIDI keyboard to mute or solo individual drum sounds in real time.

- **The keys C2 to E3 (white keys only) will mute individual drum channels starting with channel 1.**
 The sounds are muted for as long as you hold the key(s) down.
- **The keys C4 to E5 (white keys only) will solo individual drum channel, starting with channel 1.**
 The sounds are soloed for as long as you hold the key(s) down.



This is a great way to bring drum sounds in and out of the mix when playing Reason live. You can also record the drum channel Mutes in the main sequencer, just like any other controller (see "Recording parameter automation").

The Effect Sends (S1 & S2)



On the back panel of Redrum you will note two audio connections labeled "Send Out" 1 and 2. When you create a Redrum device, these will by default be auto-routed to the first two "Chaining Aux" inputs on the Mixer device (provided that these inputs aren't already in use).

This feature allows you to add effects to independent drum sounds in the Redrum.

- **Raising the S1 knob for a drum sound channel will send the sound to the first send effect connected to the mixer.**
 Similarly, the S2 knob governs the send level to the second send effect in the mixer.
- **Note that there must be send effects connected to the AUX Sends and Returns in the mixer for this to work.**
- **Also note that if Redrum is soloed in the Mixer the effect sends will be muted.**
- **Another way to add independent effects to drum sounds is to use the independent drum outputs.**
 See "Connections".

Pan



Sets the Pan (stereo position) for the channel.

- **If the LED above the Pan control is lit, the sound uses a stereo sample.**
 In that case, the Pan control serves as a stereo balance control.

Level and Velocity



The Level knob sets the volume for the channel. However, the volume can also be affected by velocity (as set with the Dynamic value, or as played via MIDI). How much the volume should be affected by velocity is set with the "Vel" knob.

- **If the Vel knob is set to a positive value, the volume will become louder with increasing velocity values.**
 The higher the Vel value, the larger the difference in volume between low and high velocity values.
- **A negative value inverts this relationship, so that the volume decreases with higher velocity values.**
- **If the Vel knob is set to zero (middle position), the sound will play at a constant volume, regardless of the velocity.**
 When Vel is set to zero, the LED above the knob goes dark.

Length and the Decay/Gate switch



The Length knob determines the length of the drum sound, but the result depends on the setting of the Decay/Gate switch:

- **In Decay mode (switch down), the sound will decay (gradually fade out) after being triggered. The decay time is determined by the Length setting.**
 In this mode, it doesn't matter for how long a drum note is held (if played back from the main sequencer or via MIDI) - the sound will play the same length for short notes as for long notes. This is the traditional "drum machine" mode.

→ **In Gate mode (switch up), the sound will play for the set Length, and then be cut off.**
Furthermore, if a sound set to Gate mode is played from the main sequencer, from a CV/Gate device or via MIDI, the sound will be cut off when the note ends or after the set Length, depending on which comes first. Or in other words, the sound plays for as long as you hold the note, but the Length setting serves as the maximum length for the sound.
There are several uses for the Gate mode:

- **For "gated" drum sounds, when the tail of the sound is abruptly cut off as an effect.**
- **For when you want to use very short sounds, and don't want them to "lose power" by being faded out.**
- **For when you play the Redrum from the sequencer or via MIDI, with sounds for which the length is important (e.g. when using the Redrum as a sound effects module).**

! **Audio samples sometimes contain a "loop", which is set by editing the audio in a sample editor. This loop repeats a part of the sample to produce sustain as long as you hold down a note. Drum samples usually don't contain loops, but who is to say that Redrum should only play drum samples?**
Note that if a sample contains a loop, and Length is set to maximum, the sound will have infinite sustain, in other words it will never become silent, even if you stop playback. Decreasing the Length setting solves this problem.

Pitch



Sets the pitch of the sound. The range is +/- 1 octave.

→ **When the pitch is set to any other value than 0, the LED above the knob lights up to indicate that the sample isn't played back at its original pitch.**

Pitch Bend



By setting the Bend knob to a positive or negative value, you specify the start pitch of the sound (relative to the Pitch setting). The pitch of the sound will then be bent to the main Pitch value. Thus, selecting a positive Bend value will cause the pitch to start higher and bend down to the original Pitch, and vice versa.

→ **The Rate knob determines the bend time - the higher the value, the slower the bend.**

→ **The Vel knob determines how the Bend amount should be affected by velocity.**
With a positive Vel value, higher velocity results in wider pitch bends.

→ **The Bend and Vel knobs have LEDs that light up when the functions are activated (i.e. when a value other than zero is selected).**

! **Pitch bend is available for drum sound channels 6 and 7 only.**

Tone



The Tone knob determines the brightness of the drum sound. Raising this parameter results in a brighter sound. The Vel knob determines whether the sound should become brighter (positive Vel value) or darker (negative Vel value) with higher velocity.

- → **The Tone and Vel knobs have LEDs that light up when the functions are activated (i.e. when a value other than zero is selected).**
- ! **The Tone controls are available for drum sound channels 1, 2 and 10 only.**

Sample Start



The Start parameter allows you to adjust the start point of the sample. The higher the Start value, the further the start point is moved "into" the sample. If you set the Start Velocity knob to a positive amount, the sample start point is moved forward with higher velocities. A negative Start Velocity amount inverts this relationship.

- → **When Start Velocity is set to any other value than zero, the LED above the knob lights up.**
- → **A negative Start Velocity amount is only useful if you have set the Start parameter to a value higher than 0.** By raising the Start value a bit and setting Start Velocity to a negative value, you can create rather realistic velocity control over some drum sounds. This is because the very first transients in the drum sound will only be heard when you play hard notes.
- ! **The Sample Start settings are available for drum sound channels 3-5, 8 and 9.**

Global settings

Channel 8 & 9 Exclusive



If this button is activated, the sounds loaded into drum channels 8 and 9 will be exclusive. In other words, if a sound is played in channel 8 it will be silenced the moment a sound is triggered in channel 9, and vice versa.

The most obvious application for this feature is to "cut off" an open hi-hat with a closed hi-hat, just like a real one does.

High Quality Interpolation



When this is activated, the sample playback is calculated using a more advanced interpolation algorithm. This results in better audio quality, especially for drum samples with a lot of high frequency content.

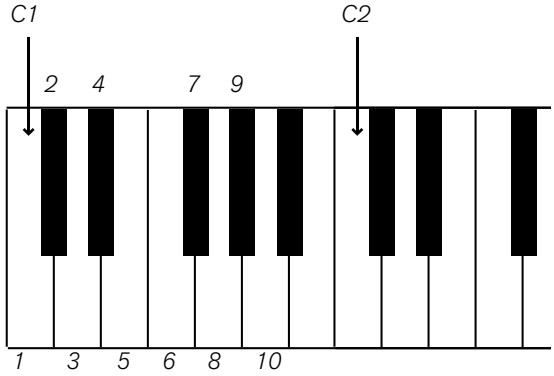
→ **High Quality Interpolation uses more computer power - if you don't need it, it's a good idea to turn it off!** Listen to the drum sounds in a context and determine whether you think this setting makes any difference.

Master Level

The Master Level knob in the top left corner of the device panel governs the overall volume from Redrum.

Using Redrum as a sound module

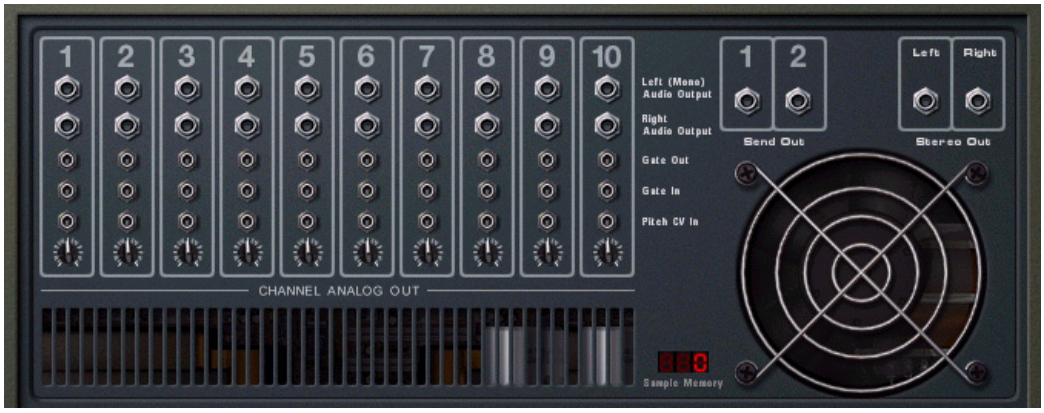
The drum sounds in Redrum can be played via MIDI notes. Each drum sound is triggered by a specific note number, starting at C1 (MIDI note number 36):



This allows you to play Redrum live from a MIDI keyboard or a MIDI percussion controller, or to record or draw drum notes in the main sequencer. If you like, you can combine pattern playback with additional drum notes, such as fills and variations. However:

! **If you want to use Redrum purely as a sound module (i.e. without pattern playback) you should make sure that the "Enable Pattern Section" button is deactivated (see "The Enable Pattern Section switch"), otherwise the Redrum pattern sequencer will start as soon as you start the main sequencer.**

Connections



On the back of the Redrum you will find the following connections:

For each drum sound channel:

Connection	Description
Audio Outputs	There are individual audio outputs for each drum sound channel, allowing you to route a drum sound to a separate channel in the mixer, possibly via insert effects, etc. For mono sounds, use the "Left (Mono)" output (and pan the sound using the Pan control in the mixer). When you use an individual output for a sound, the sound is automatically excluded from the master stereo output.
Gate Out	This sends out a gate signal when the drum sound is played (from a pattern, via MIDI or by using the Trigger button on the device panel). This lets you use the Redrum as a "trig sequencer", controlling other devices. The length of the gate signal depends on the Decay/Gate setting for the sound: In Decay mode, a short "trig pulse" is sent out, while in Gate mode, the gate signal will have the same length as the drum note.
Gate In	Allows you to trigger the sound from another CV/Gate device. All settings apply, just as when playing the drum sound conventionally.
Pitch CV In	Lets you control the pitch of the drum sound from another CV device.

Other

Connection	Description
Send Out 1-2	Outputs for the send signals controlled with the S1 and S2 knobs.
Stereo Out	This is the master stereo output, outputting a mix of all drum sounds (except those for which you use individual outputs).

Chapter 12
Dr. Octo Rex
Loop Player



Introduction



The Dr. Octo Rex Loop Player is the successor to the trusty Dr. Rex Loop Player, introduced in Reason Version 1. The Dr. Octo Rex can hold up to eight different REX loops at once, in eight pattern memories, and allows you to switch between loops and slices in very flexible ways.

The Dr. Octo Rex Loop Player is capable of playing back and manipulating files created in ReCycle, another product created by Reason Studios, or bounced from open Single Take audio clips in the stand-alone version of Reason. ReCycle is a program designed especially for working with sampled loops. By “slicing” an audio loop and making separate samples of each beat, ReCycle makes it possible to change the tempo of loops without affecting the pitch and to edit the loop as if it was built up of individual sounds.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

ReCycled Loops

To fully understand Dr. Octo Rex you need to understand what it means to ReCycle a drum loop. Imagine that you have a sample of a drum loop that you want to use in a track you are working on. The loop is 144 BPM and your track is 118 BPM. What do you do? You can of course lower the pitch of the loop, but that will make the loop sound very different, and if the loop contains pitched elements they will no longer match your song. You can also time stretch it. This won’t alter the pitch, but will make the loop sound different. Usually it means that you loose some “punch” in the loop.

Instead of stretching the sample, ReCycle slices the loop into little pieces so that each drum hit (or whatever sound you are working with) gets its own slice. These slices can be exported to an external hardware sampler or saved as a REX file to be used in Reason. When the loop has been sliced you are free to change the tempo any way you want.

About REX file formats

Dr. Octo Rex can read REX files in the following formats:

- **REX (.rex)**
 This is the file format generated by previous versions of ReCycle (Mac platform).
- **RCY (.rcy)**
 This is the file format generated by previous versions of ReCycle (PC platform).
- **REX 2 (.rx2)**
 This is the ReCycle file format for both Mac and PC platforms generated internally in the stand-alone version of Reason or by ReCycle version 2.0 and later. One of the differences between the original REX format and REX2, is that the REX2 format supports stereo files.

! **Unlike the Dr. Rex device, Dr. Octo Rex can also load and save the device panel settings in a special Patch format (.drex). The REX file(s) and the Dr. Octo Rex panel settings are also saved in the Song file just like every other patch in the song.**

Loading and saving Dr. Octo Rex patches

Loading and saving Dr. Octo Rex patches is done in the same way as in other patch based devices - see "Loading patches" and "Saving patches".

Dr. Octo Rex patches can consist of up to eight separate REX loops. When you load an Dr. Octo Rex patch, the REX loop(s) will be automatically loaded in the Loop Slots with their names shown in the display(s) below each button.

- **You can also load Dr. Octo Rex patches by dragging them from the Browser and dropping on the device panel.**



About the Dr. Octo Rex patch format

Dr. Octo Rex patches (.drex) contains all panel and synth parameter settings as well as references to all (up to eight) REX loops. The actual REX files are not contained in the patch but must be available separately on the computer.

Playing Loops

1. **Make sure the Enable Loop Playback button is on (lit).**



2. **Click the desired Loop Slot button.**
3. **Play back the loop by clicking the Run button.**
 The loop in the selected Loop Slot will play back repeatedly in the tempo set in the main sequencer. If you change the tempo, the loop tempo will follow.

- **You can also play the loop once via MIDI, by using the D0 key.**
- **To check out the loop(s) together with other device sequencer data and patterns already recorded, click the sequencer Play button.**
 The loops will automatically play back in perfect sync with the sequencer.

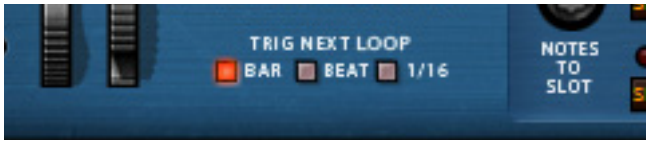
Switching playback between Loop Slots

Switching playback between loops in different Loop Slots is just like switching a Pattern in a Redrum device, for example.

1. **Activate the Enable Loop Playback button on the Dr. Octo Rex device.**
2. **Click the Run button - or start sequencer playback.**
3. **Click another Loop Slot button to switch loop.**
 Selecting Loop Slots that have no loops loaded will result in silence.

The Trig Next Loop function

The Trig Next Loop function determines how long Dr. Octo Rex waits after a Loop Slot button (or a key) is pressed before it actually "gates in" or triggers the loop. This allows for different "precision" when switching between running loops:



- **Activate the Bar button to make the loops switch at the next bar of the current loop.**
- **Activate the Beat button to make the loops switch at the next beat of the current loop.**
- **Activate the 1/16 button to make the loops switch at the next 1/16th note of the current loop.**

Switching Loops using Pattern Automation in the sequencer

Switching between Loop Slots can be automated, by using parameter automation in the main sequencer.

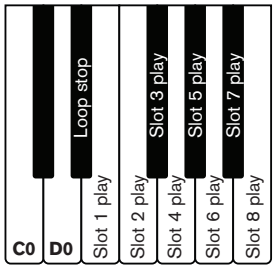
! **When using parameter automation, the Trig Next Loop function described above is disregarded - the switching between Loop Slots is instantaneous.**

Triggering playback and selecting Loop Slots from a MIDI keyboard

It's also possible to control playback, stop and Loop Slot selection in real-time by pressing different keys on a MIDI keyboard. By pressing the keys E0 to B0 you select Loop Slot 1-8 and start playback of the corresponding loop. The loop(s) will play back continuously, one loop at a time, until you press the D#0 key to stop playback, or click the Run button or Stop in the main sequencer. The time between key press and Loop Slot switching is determined by the Trig Next Loop function, see "The Trig Next Loop function".

! **Note that the Enable Loop Playback button must be on.**

The picture shows what keys should be pressed to select and play Loop Slot and to stop loop playback:



- **To maintain backwards compatibility with Dr. Rex, the D0 key can be used to play back the REX loop in the Loop Slot that currently has Note To Slot focus (see "Note To Slot").**
 The loop is played back once (single-shot) and cannot be stopped during this time.

Adding Loops

To add one or several (max 8) loops into the Dr. Octo Rex Loop Player, proceed as follows:

1. **Unfold the Loop Editor panel.**
2. **Select the Loop Slot you wish to add the (first) REX loop into.**



3. **Open the REX Loop browser by clicking the folder button to the left of the Loop Slot buttons.**
 Alternatively, select "Browse Loops..." from the Edit menu or the device context menu.
4. **In the Browser, locate and select the desired loop(s).**
 You can listen to the loops before loading by using the Preview function in the Browser.

→ **To select several loops, hold down [Ctrl](Win)/[Cmd](Mac) and click.**
To select a range of loops, hold down [Shift] while clicking the last file.

5. **Click the Load button in the Browser to load the selected file(s) in the Loop Slot(s).**

! **If you have selected and opened several loops, the first loop will load in the selected Loop Slot and the rest will load in consecutive Loop Slots.**

! **Loading new REX files will replace any files currently in the slots.**

→ **Alternatively, select the REX loop(s) in the Browser and drag and drop it/them on the Loop Editor panel section, or on a Loop Slot button on the Controller Panel.**
If you have selected several loops, the first loop will load in the selected Loop Slot and the rest will load in consecutive Loop Slots.

! **If you drag a single REX loop from the Browser and drop on the Controller Panel (not on a Loop Slot button), the REX loop will load into Slot 1 and all other Slots will be cleared.**

Loading Loops "On the Fly"

Another practical method for checking out loops, is to load them "on the fly", i.e. during playback. This is especially useful if you want to check out a number of loops against other sequencer data and patterns previously recorded. Proceed as follows:

1. **Activate the Enable Loop Playback button on the Dr. Octo Rex device and start sequencer playback.**
 The REX loops and the sequencer are synced.
2. **Now load a new REX file by using the Browser in one of the usual ways.**
 After a brief silence, the new file is loaded, and sync is maintained.
3. **Repeat step 2 as necessary until you have found a suitable loop.**

→ **If you are trying out loops within the same folder, the quickest ways to select and load a new loop is to use the arrow buttons next to the loop name display.**
Or, you can click in the loop name display and select a new loop from the pop-up menu that appears.

Removing Loops

→ **To remove a loop from a Loop Slot, select "Remove Loop" from the Edit menu or device panel context menu.**

Cut/Copy and Paste Loops between Loop Slots

To cut or copy a loop from one Loop Slot and paste into another, proceed as follows:

1. **Click the button of the Loop Slot that contains the loop you want to cut or copy.**
2. **Select "Cut Loop" or "Copy Loop" from the panel context menu.**

! **You have to context-click on the panel (not any button or knob) to access the correct context menu.**

3. **Click the destination Loop Slot button and select "Paste Loop" from the panel context menu.**
 Now, you can edit the slices of the pasted loop as desired, see "Slice handling".

Playing individual Loop Slices

Besides playing back entire REX Loops using the Run function (or Play in the sequencer), it's also possible to play individual slices of a loop from a MIDI master keyboard. This way you can use the Dr. Octo Rex almost like a traditional sampler, playing separate slices from separate keys.

The slices are automatically distributed in semitone steps, with the first slice on MIDI note C1, the second slice on C#1 and so on, with one note for each slice. The note range differs depending on how many slices the REX Loop contains.

→ **Define which REX Loop to control from the MIDI master keyboard by selecting the desired Loop Slot with the Note To Slot knob:**



The range is 1-8 corresponding to Loop Slots 1-8. Selected Slot is indicated with a lit LED.

Slice handling

Selecting Slices



A selected slice is indicated by being highlighted in the waveform display. To select a slice, use one of the following methods:

- **By clicking in the waveform display.**
 If you hold down [Alt](Win) or [Option](Mac) and click on a slice in the waveform display, it will be played back. The pointer takes on the shape of a speaker symbol to indicate this.
- **By using the "Slice" knob below the waveform display.**
- **Via MIDI.**
 If you activate "Select Slice Via MIDI", you can select and "play" slices using your MIDI keyboard. Slices are always mapped to consecutive semitone steps, with the first slice always being on the "C1" key.



- **If you play back a loop with "Select Slice via MIDI" option activated, each consecutive slice is selected in the waveform display as you play the keys.**
 You can edit parameters during playback.

Editing Slices in the Waveform Display



Here you are able to edit several parameters for each slice, by first selecting the slice and then using the knobs below the waveform display. If you want to edit a single parameter for several slices at once, a more convenient way would be to use the Slice Edit Mode, see "The Slice Edit Mode". The following slice parameters can be set:

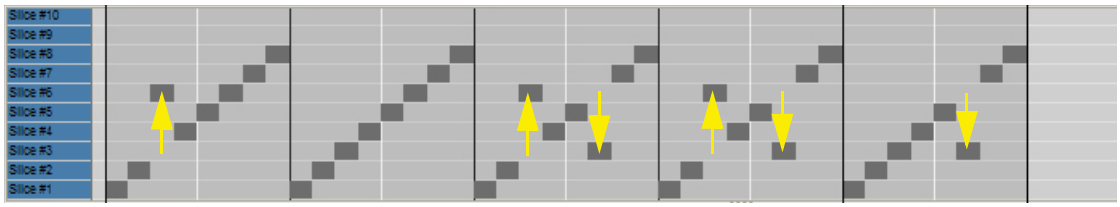
Parameter	Description
Pitch	Allows you to transpose each individual slice in semitone steps, over a range of more than eight octaves.
Pan	The stereo position of each slice.
Level	The volume of each slice. The default level is 100.
Decay	Allows you to shorten individual slices.
Rev	Allows you to play back individual slices reversed (backwards)
F.Freq	Allows you to modify the Filter (cutoff) Frequency of individual slices. This value is added to, or subtracted (if negative) from the FREQ value of the synth panel, see "Filter Frequency".
Alt	Allows you to assign slices to an Alternate group (1-4). Slices assigned to any of these four Alt groups will be played pack in a random fashion within each group, see "About the Alt parameter".
Output	Allows you to assign individual slices to separate audio outputs (1-8). If the REX loop is in stereo, there is also an option to select individual output pairs (1+2, 3+4, 5+6 or 7+8) for individual slices.

! **If you have made settings to any of the parameters listed above, these will be lost if you load a new REX file into that Loop Slot.**

About the Alt parameter

The Alt parameter in the waveform display can be used to create a more "live" feel to your Rex loops by alternating slices within each individual Alt group. For example, if you assign all snare hit slices in the loop to the Alt 1 group, the snare slices will be selected and played back randomly each time these slices appear in the loop. Then, you could assign all hi-hat slices to the Alt 2 group and so on. The result will be a loop that plays back differently every cycle.

In the example below, slices 3 and 6 have been assigned to the same Alt group. Here, we show the loop played back five times, just so you can see the slice alternation. As you can see, slices 3 and 6 have been distributed randomly for each loop cycle:



This randomization within each Alt group also occurs when you play back the REX loop using the Run function - and when you use parameter automation in the main sequencer.

The Slice Edit Mode

A very convenient way of editing several slices at once is to work in Slice Edit Mode. In Slice Edit Mode, you can edit one parameter at a time for all slices in the loop.

1. **Click the Slice Edit Mode button.**



The waveform display switches to show the REX loop in Slice Edit Mode.

2. **Select the parameter you want to edit by clicking on its name below the REX loop.**
The parameters that can be selected are: Pitch, Pan, Level, Decay, Reverse, Filter Frequency, Alt Group and Output.



Here, we have selected the Pitch parameter.

3. **Edit the Pitch value for each individual slice by clicking, or drawing across several slices, in the display.**



Now, the Pitch parameter can be edited for all slices in a single sweep.

→ **To reset the selected parameter to its default value for one or several slices, hold down [Ctrl](Win) or [Cmd](Mac) and click on the desired slice(s), or draw across the slices in the waveform display.**

4. **When you are finished with one parameter, select another parameter and repeat the procedure by drawing values for the slices in the waveform display.**

! **If you have made settings to any of the parameters listed above, these will be lost if you load a new REX file into that Loop Slot.**

Dr. Octo Rex panel parameters

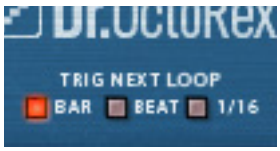
Pitch and Mod wheels



The Pitch wheel to the left is used for “bending” the pitch up or down. The Mod wheel can be used to apply various modulation while you are playing the loop(s). Virtually all MIDI keyboards have Pitch Bend and Modulation controls. Dr. Octo Rex also has two “wheels” on the panel that could be used to apply real time modulation and pitch bend should you not have these controllers on your keyboard, or if you aren’t using a keyboard at all. The wheels mirror the movements of the corresponding MIDI keyboard controllers.

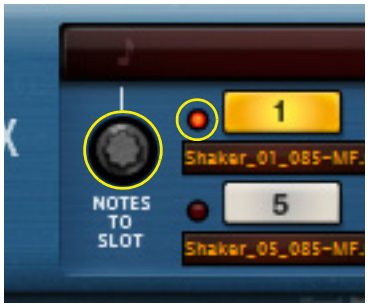
The Pitch bend range and Mod destination parameters are set on the synth parameter panel, see “Pitch Bend Range” and “Mod. Wheel”.

Trig Next Loop



The Trig Next Loop parameter determines the timing when switching between Loop Slots See “The Trig Next Loop function”.

Note To Slot



The Notes To Slot knob set to Slot 1

The Note To Slot knob controls which Loop Slot is currently controlled from the MIDI master keyboard, or by any recorded sequencer notes. The Loop Slot which currently has “note input” is indicated with a lit LED.

The Note To Slot parameter can also be automated in the main sequencer. This means you could switch between Loop Slots for every single sequencer note if you like. This opens up for very interesting “beat mangling” experiments.

Loop Slot buttons

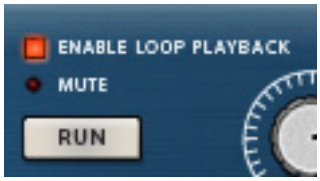


The eight Loop Slot buttons are located in the center of the front panel. You can load one REX loop per Slot. Loading REX loops are done from the Loop Editor panel, see "Select Loop & Load Slot".

→ **Click a Loop Slot button to select its REX loop for playback.**
Play back the REX loop in the selected Loop Slot by clicking the Run button (or Play in the main sequencer).

! **Note that selecting a Loop Slot only selects the corresponding REX loop for playback using the Run function (see "Enable Loop Playback and Run") or Play from the main sequencer. Which Loop Slot the master keyboard or sequencer notes control is defined with the Note To Slot button, see "Note To Slot".**

Enable Loop Playback and Run



→ **Click the Enable Loop Playback button to make it possible to play back the REX loops using the Run button or Play function in the main sequencer.**
If the Enable Loop Playback button is off, clicking Run or Play in the sequencer won't play back the loops. This can be useful if you only want to control the individual slices of the REX loops from a master keyboard or from recorded notes in the main sequencer.

Volume



The Master Volume parameter acts as a general volume control for the loops in all Loop Slots.

Global Transpose



Set the global transposition of the loops in all Loop Slots by using the Global Transpose spin control. You can raise or lower the pitch in 12 semitone steps (+/– 1 octave).

- **The Global Transpose value can also be controlled via MIDI, by pressing a key between C-2 and C0 (with C-1 resetting the transpose value to zero).**
 This way you can also record transposition changes in the sequencer.

Dr. Octo Rex synth parameters

The Dr. Octo Rex synth parameters are used for shaping and modulating the sound of the REX loops. These parameters are familiar synth parameters, similar to the ones in the synthesizers; The Subtractor and the Malström, and in the samplers; the NN-19 and the NN-XT. It is important to remember that these parameters do not alter the REX files in any way, only the way they will play back.

! **Most of the synth parameters are global, in the sense that they will affect all slices in the REX files as well as all REX loops in all eight Loop Slots.**

! **All Dr. Octo Rex synth panel settings are stored in the Song (and in the Dr. Octo Rex patch file if you choose to save the settings as a patch).**

Select Loop & Load Slot



→ **Click any of the eight Select Loop & Load Slot buttons to select a loaded REX loop for editing, or to load a new REX file to.**
 If no loop is already present in the selected Loop Slot, the Waveform Display will be blank. Otherwise, the display shows a graphical readout of the REX loop and info (name, original loop tempo, number of bars and signature).

→ **Click the Follow Loop Playback button to "synchronize" the Select Loop & Load Slot buttons to the Loop Slot buttons on the front panel.**
 This way, the currently playing loop will always be displayed in the Waveform Display. If you're using Pattern Automation in the sequencer, where the Slots are switched during playback, you might want to deactivate the Follow Loop Playback function to make it easier to edit a specific loop.

Refer to "Adding Loops" for info on how to load REX files and to "Editing Slices in the Waveform Display" for info about editing the REX loop.

Copy MIDI Notes to a sequencer track

It's possible to copy the MIDI Notes of the REX loop of the currently selected Slot to a track in your DAW's sequencer:

1. **Click to select the desired Loop Slot.**
2. **Click the “Drag MIDI Notes to track in sequencer” icon and drag to the destination track in the sequencer:**



! **Be sure to disable the Enable Loop Playback function on the Dr Octo Rex panel to avoid “doubled notes”.**

Loop Transpose



→ **Set the transposition of individual loops in the Dr. Octo Rex by using the Loop Transpose knob to the bottom left on the panel, or by clicking on the keyboard display below the knob.**
You can raise or lower the pitch in 12 semitone steps (+/−1 octave).

← **It’s also possible to set a global transpose value that affects all REX loops equally, see “Global Transpose”.**

Loop Level

→ **Set the individual levels for the loops in the Loop Slots with the Loop Level knob.**
This lets you match the levels of the loops in the 8 Loop Slots.

Oscillator section



For a REX file, the audio contained in the slices are what oscillators are for a synthesizer, the main sound source. The following settings can be made in the Osc Pitch section of the Dr. Octo Rex:

Env. A

This parameter determines to what degree the overall pitch of all the REX files will be affected by the Filter Envelope (see "Filter Envelope"). You can set negative or positive values here, which determines whether the envelope curve should raise or lower the pitch.

Oct and Fine - Setting the overall Pitch

You can change the overall pitch of all REX files in the 8 Loop Slots in three ways:

- **In octave steps.**
 This is done using the Oct knob. The range is 0 - 8, with "4" as default.
- **In semitone steps.**
 This is done by using Global Transpose controls, see "Global Transpose".
- **In cents (hundredths of a semitone).**
 The range is -50 to 50 (down or up half a semitone).

→ **To transpose individual REX loops, use the Loop Transpose parameter, see "Loop Transpose".**

! **To tune an individual slice in a REX loop, select it and use the Pitch parameter below the waveform display, see "Editing Slices in the Waveform Display".**

Mod. Wheel



The Modulation wheel can be set to simultaneously control a number of parameters. You can set positive or negative values, just like in the Velocity Control section. The following parameters can be affected by the modulation wheel:

Parameter	Description
F. Freq	This sets modulation wheel control of the filter frequency parameter. A positive value will raise the frequency if the wheel is pushed forward. Negative values invert this relationship.
F. Res	This sets modulation wheel control of the filter resonance parameter. A positive value will increase the resonance if the wheel is pushed forward. Negative values invert this relationship.
F. Decay	This sets modulation wheel control for the Filter Envelope Decay parameter. A positive value will increase the decay if the wheel is pushed forward. Negative values invert this relationship.

Velocity section



Velocity is usually used to control various parameters according to how hard or soft you play notes on your keyboard. A REX file does not contain velocity values on its own. As velocity information is meant to reflect variation, having them all set to the same value is not meaningful if you wish to velocity control Dr. Octo Rex parameters.

There are basically two ways you can apply "meaningful" velocity values to REX files:

- **After having recorded MIDI notes in the main sequencer, you can edit the velocity values in the in the sequencer.**
- **You can play slices in real time on your keyboard. The resulting data will have velocity values reflecting how the notes were struck when you played.**

When velocity values have been adjusted, you can control how much the various parameters will be affected by velocity. The velocity sensitivity amount can be set to either positive or negative values, with the center position representing no velocity control.

The following parameters can be velocity controlled:

Parameter	Description
F. Env	This sets velocity control for the Filter Envelope Amount parameter. A positive value will increase the envelope amount with higher velocity values. Negative values invert this relationship.
F. Decay	This sets velocity control for the Filter Envelope Decay parameter. A positive value will increase the Decay time with higher velocity values. Negative values invert this relationship.
Amp	This let's you velocity control the overall volume of the file. If a positive value is set, the volume will increase with higher velocity values.

The Filter Section



Filters are used for shaping the overall timbre of all REX files in all 8 Loop Slots. The filter in Dr. Octo Rex is a multimode filter with five filter modes.

→ **Activate or deactivate the filter completely by clicking the Filter On button.**
The filter is active when the button is lit.

Mode

With this selector you can set the filter to operate as one of five different types of filter. These are as follows:

- **Notch**
A notch filter (or band reject filter) could be described as the opposite of a bandpass filter. It cuts off frequencies in a narrow midrange band, letting the frequencies below and above through.
- **High-Pass (HP12)**
A highpass filter is the opposite of a lowpass filter, cutting out lower frequencies and letting high frequencies pass. The HP filter slope has a 12 dB/Octave roll-off.
- **Bandpass (BP 12)**
A bandpass filter cuts both high and low frequencies, while midrange frequencies are not affected. Each slope in this filter type has a 12 dB/Octave roll-off.
- **12 dB Lowpass (LP 12)**
This type of lowpass filter is also widely used in classic analog synthesizers (Oberheim, early Korg synths, etc.). It has a gentler slope (12 dB/Octave), leaving more of the harmonics in the filtered sound compared to the LP 24 filter.

- **24 dB Lowpass (LP 24)**
 Lowpass filters lets low frequencies pass and cuts out the high frequencies. This filter type has a fairly steep roll-off curve (24dB/Octave). Many classic synthesizers (Minimoog/Prophet 5 etc.) used this filter type.

Filter Frequency

The Filter Frequency parameter (often referred to as "cutoff") determines which area of the frequency spectrum the filter will operate in. For a lowpass filter, the frequency parameter could be described as governing the "opening" and "closing" of the filter. If the Filter Freq is set to zero, none or only the very lowest frequencies are heard, if set to maximum, all frequencies in the waveform are heard. Gradually changing the Filter Frequency produces the classic synthesizer filter "sweep" sound.

! **Note that the Filter Frequency parameter is usually controlled by the Filter Envelope (see "Filter Envelope") as well. Changing the Filter Frequency with the Freq slider may therefore not produce the expected result.**

Resonance

The filter resonance parameter affects the character of the filter sound. For lowpass filters, raising the resonance will emphasize the frequencies around the set filter frequency. This produces a generally thinner sound, but with a sharper, more pronounced filter frequency "sweep". The higher the resonance value, the more resonant the sound becomes until it produces a whistling or ringing sound. If you set a high value for the resonance parameter and then vary the filter frequency, this will produce a very distinct sweep, with the ringing sound being very evident at certain frequencies.

- **For the highpass filter, the resonance parameter operates just like for the lowpass filters.**
- **When you use the Bandpass or Notch filter, the resonance setting adjusts the width of the band.**
 When you raise the resonance, the band where frequencies are let through (Bandpass), or cut (Notch) will become narrower. Generally, the Notch filter produces more musical results using low resonance settings.

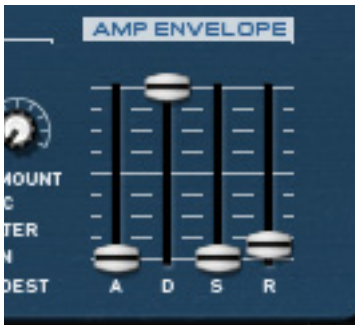
Envelope section

Envelope generators are used to control several important sound parameters in analog synthesizers, such as pitch, volume, filter frequency etc. In a conventional synthesizer, envelopes govern how these parameters should respond over time - from the moment a note is struck to the moment it is released. In the Dr. Octo Rex device however, the envelopes are triggered each time a slice is played back.

There are two envelope generators in the Dr. Octo Rex, one for volume, and one for the filter frequency (and/or pitch). Both have the standard four parameters; Attack, Decay, Sustain and Release

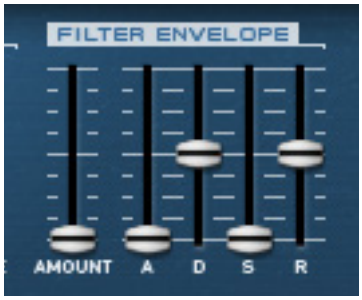
! **Please refer to "Envelopes - General" in the Subtractor chapter for a description of the basic envelope parameters.**

Amplitude Envelope



The Amp Envelope governs how the volume of each slice should change over time, from the time it is triggered (the slice note starts) until the slice note ends. This can be used to make a loop more distinct (by having a snappy attack and a short decay time) or more spaced-out (by raising the attack time).

Filter Envelope



The Filter Envelope can be used to control two parameters for all REX loops in the 8 Loop Slots; filter frequency and overall loop pitch. By setting up a filter envelope you control how the filter frequency and/or the pitch should change over time for each slice.

The Amount parameter determines to what degree the filter frequency will be affected by the Filter Envelope. The higher the Amount setting, the more pronounced the effect of the envelope on the filter.

- **Try lowering the Frequency slider and raising Resonance and Envelope Amount to get the most effect of the filter envelope!**

LFO section



LFO stands for Low Frequency Oscillator. LFOs are oscillators in the sense that they generate a waveform and a frequency. However, there are two significant differences compared to normal sound generating oscillators:

- **LFOs only generate waveforms with low frequencies.**
- **The output of the two LFOs are never actually heard. Instead they are used for modulating various parameters.**

The most typical application of an LFO is to modulate the pitch of a (sound generating) oscillator or sample, to produce vibrato. In the Dr. Octo Rex device, you can also use the LFO to modulate the filter frequency or panning.

Waveform

LFO 1 allows you to select different waveforms for modulating parameters. These are, from top to bottom:

Waveform	Description
Triangle	This is a smooth waveform, suitable for normal vibrato.
Inverted Sawtooth	This produces a "ramp up" cycle. If set to control pitch (frequency), the pitch would sweep up to a set point (governed by the Amount setting), after which the cycle immediately starts over.

Waveform	Description
Sawtooth	This produces a “ramp down” cycle, the same as above but inverted.
Square	This produces cycles that abruptly changes between two values, usable for trills etc.
Random	Produces random stepped modulation to the destination. Some vintage analog synths called this feature “sample & hold”.
Soft Random	The same as above, but with smooth modulation.

Destination

The available LFO Destinations are as follows:

Destination	Description
Osc	Selecting this makes LFO control the pitch (frequency) of the REX file.
Filter	Selecting this makes the LFO control the filter frequency.
Pan	Selecting this makes the LFO modulate the pan position of the REX file, i.e. it will move the sound from left to right in the stereo field.

Sync

By clicking the SYNC button you activate/deactivate LFO sync. The frequency of the LFO will then be synchronized to the main sequencer tempo, in one of 16 possible time divisions. When sync is activated, the Rate knob (see below) is used for setting the desired time division.

Turn the knob and check the tooltip for an indication of the time division.



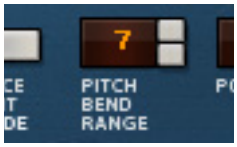
Rate

The Rate knob controls the LFO's frequency. Turn clockwise for a faster modulation rate.

Amount

This parameter determines to what degree the selected parameter destination will be affected by the LFO 1, i.e. the amount of vibrato, filter wah or auto-panning.

Pitch Bend Range



The Pitch Bend Range parameter sets the amount of pitch bend when the wheel is turned fully up or down. The maximum range is 24 semitones (=up/down 2 Octaves).

Setting number of voices - polyphony



This determines the polyphony, i.e. the number of voices, or slices, Dr. Octo Rex can play simultaneously. For normal loop playback, it is worth noting that slices sometimes "overlap". Therefore, it is recommended that you use a polyphony setting of about 3-4 voices when playing REX files. If you are "playing" slices via MIDI, the polyphony setting should be set according to how many overlapping slices you want to have.

! **Note that the Polyphony setting does not "hog" voices. For example, if you are playing a file that has a polyphony setting of ten voices, but the file only uses four voices, this doesn't mean that you are "wasting" six voices. In other words, the polyphony setting is not something you need to consider if you want to conserve CPU power - it is only the number of voices actually used that counts.**

Audio Quality settings

Dr. Octo Rex features two parameters that provide ways of balancing audio quality vs. conservation of computer power. The parameters are called "High Quality Interpolation" and "Low Bandwidth" and are located to the right on the rear panel:



High Quality Interpolation

When High Quality Interpolation is active, the loop file playback is calculated using a more advanced interpolation algorithm. This results in better audio quality, especially for loops with a lot of high frequency content.

- **High Quality Interpolation uses more computer power - if you don't need it, it's a good idea to turn it off.** Listen to the loop in a context and determine whether you think this setting makes any difference.

Low Bandwidth

This will remove some high frequency content from the sound, but often this is not noticeable (this is especially true if you have "filtered down" your loop). Activating this mode will save you some extra computer power, if needed.

Connections



On the rear panel of Dr. Octo Rex you will find the connectors. The left part of the panel houses a number of CV/Gate inputs and outputs. Using CV/Gate is described in "Routing Audio and CV".

Modulation Inputs

These control voltage (CV) inputs (with trim pots), allow you to modulate various Dr. Octo Rex parameters from other devices (or from the modulation outputs of the Dr. Octo Rex device itself). The following CV inputs are available:

- **Master Volume**
- **Mod Wheel**
- **Pitch Wheel**
- **Filter Cutoff**
- **Filter Resonance**
- **Osc Pitch**

Modulation Outputs

The Modulation outputs can be used to voltage control other devices, or other parameters in the Dr. Octo Rex device itself. The Modulation Outputs are:

- **Filter Envelope**
 The Filter Envelopes in Dr. Octo Rex are polyphonic (one per voice). Only the filter envelope of voice 1 is output here.
- **LFO**

Gate Inputs

These inputs can receive a CV/gate signal to trigger the two envelopes. Note that connecting to these inputs will override the "normal" triggering of the envelopes. For example, if you connected an LFO CV output on another device to the Gate Amp input on the Dr. Octo Rex, the amplitude envelope would not be triggered by the incoming MIDI notes to the Dr. Octo Rex device, but by the LFO CV signal. In addition you would only hear the LFO triggering the envelope for the slices that were playing at the moment of the trigger.

- **Amp Envelope**
- **Filter Envelope**

Gate Output

This outputs a gate signal for each triggered slice in the loop.

Slice Outputs

To the right of the modulation inputs and outputs are the eight individual slice audio outputs. You can assign individual slices to any of these outputs as described in "Editing Slices in the Waveform Display".

Main Outputs

To the right are the main left and right audio outputs. When you create a new Dr. Octo Rex device, these are auto-routed to the first available outputs in the I/O device.



Chapter 13
Europa
Shapeshifting
Synthesizer

Introduction



The Europa Shapeshifting Synthesizer is the most advanced and sonically "wide" synthesizer in Reason. Despite being a very advanced synthesizer, it's really easy to create great sounds from scratch. Just a few mouse clicks and knob twists in a Sound Engine section will generate truly impressive and inspiring sounds!

The three powerful and flexible sound engines offer a unique combination of analog/wavetable/spectral/physical modeling/FM synthesis techniques. If you like, you could also draw your own waveforms and filter curves to design your own completely unique sounds. In addition to this you can also load your own sample into Europa and use as a wavetable and/or filter! Each sound engine also has its own Unison module for generating really wide multi-voice stereo chorus effects.

The extensive Envelopes section and Modulation Bus section allow for very detailed and flexible modulation and control. Europa also features a top-notch semi-modular multi-effect section so you could put that final touch on your sounds.

Don't forget to check out the Europa videos here!

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

Panel overview

The Europa front panel contains the following sections:

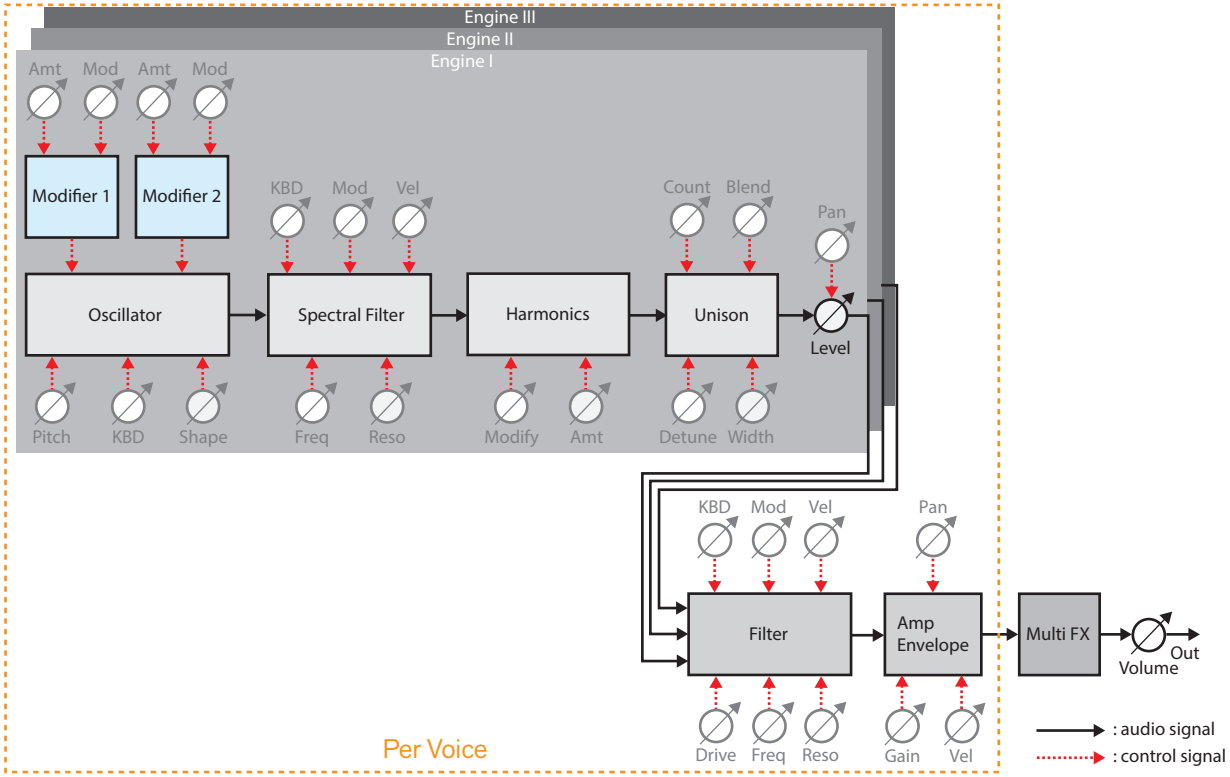


The Europa front panel sections.

- **1. MIDI Note On LED.**
- **2. Patch Selector (for browsing, loading and saving patches).**
- **3. Sound Engines section.**
- **4. User Wave and Mixer section.**
- **5. Filter section.**
- **6. Amplifier section.**
- **7. Global output controls.**
- **8. Global performance and "play" controls.**
- **9. Envelopes section.**
- **10. LFO section.**
- **11. Modulation Bus section.**
- **12. Effects section.**

Signal flow

The picture below shows the basic signal flow in Europa:



Europa signal flowchart.

- **The "hearts" of Europa are the three Sound Engines I, II and III.**
- **First in each Sound Engine is an Oscillator, which generates the basic audio signal.**
 The oscillators in Europa are extremely powerful and flexible. Besides all the basic "analog" waveforms, the oscillators can also generate a vast variety of wavetable waveforms, physical modeling signals and other types of unique signals - and also your own samples! The signals can also be continuously transformed into various shapes.
- **The Oscillator signal can be modified by the two Modifiers.**
 The Modifiers feature a huge amount of algorithms that can modify the Oscillator signal in various ways.
- **The signal from the Oscillator is routed to the Spectral Filter.**
 The Spectral Filter affects the partials of the signal. The algorithms could be various types of filters - or special purpose signal processors.
- **The signal from the Spectral Filter can then be routed to the Harmonics processor.**
 The Harmonics processor modulates the harmonics in the signal, for example introducing ensemble or stretch effects.
- **The signal from the Harmonics processor can then be routed to the Unison module.**
 This module can generate various types of unison chorus effects, to make the sound really fat and wide.
- **The signals from the Sound Engines are then routed to the Mixer, where you can set the mix between the three Sound Engine output signals and also pan the signals individually.**
- **The mixed signal is then routed to the Filter, Amp Envelope and then, via the Multi FX section, to the stereo outputs.**
- **The remaining sections in Europa (Envelopes 1-4 and LFOs 1-3) can be freely assigned to modulate destination parameters via the Modulation Bus section.**

Playing and using Europa

Loading and saving patches



Loading and saving patches is done in the same way as with any other internal Reason device, see “Loading patches” and “Saving patches” for details.

Global output controls



Master Volume

This is the main stereo output volume control.

Voices

Here you set the desired maximum polyphony of your patch, from 1 to 16 voices. This control is mainly intended for deliberately restricting the polyphony of a sound. If you just want to play a patch polyphonically you can leave the Voices setting at 16 at all times. The DSP Load won't increase with higher voice number settings - only if you play a lot of notes simultaneously.

- **If you want monophonic playback you could use the “Retrig” and “Legato” modes instead of lowering the Voices parameter to 1.**

Global performance and “play” controls



Key Mode

Here you choose how Europa should respond to MIDI Note data:

- **Poly**
 Select this if you want to play Europa polyphonically. The maximum number of voices is 16. The number of voices is set in the Voices control at the center right of the Europa panel, see "Voices".
- **Retrig**
 Select this if you want to play Europa in monophonic mode and always retrigger the envelopes as soon as you play a new note.
- **Legato**
 The Mono Legato mode is also monophonic. However, if you play a new note without having released the previous one, the envelopes won't start over.

Porta

Portamento makes note pitches glide from previous notes to new ones, at the time set with the Time knob. Portamento can be used in all Key modes (see above).

- **When On in Poly Key Mode (see above), the pitches will glide from any of the available voices.**
 The results will be unpredictable since there is no way of controlling from which note(s) the glide(s) will commence. The effect is very nice, though.
- **When On in Retrig or Legato Key Mode (see above), the pitch will glide between consecutive notes.**
- **In Auto mode, the pitch will glide between consecutive monophonic notes only when you play legato. If you have selected Poly Key Mode (see above), Auto will have no effect at all.**
 If you release the previous key before hitting the new key, there will be no portamento effect.

P.Range

→ **Set the desired Pitch Bend range for the "Pitch" wheel with the up/down buttons, or by click-holding on the display and dragging up/down.**
Range: +/-24 semitones (+/-2 octaves) in steps of +/-1 semitone.

Pitch

The Pitch bend wheel can be used for bending note pitches up and down. Europa also responds to Pitch Bend MIDI data from a connected MIDI master keyboard. You set the desired Pitch bend Range with the "P.Range" control above the Pitch bend wheel.

Mod

The Mod wheel can be used for controlling almost any parameter in Europa. Use the Mod wheel as a Source parameter in any of the Modulation Source boxes in the different sections. Or use it as Source parameter in the Modulation Bus section and then route to the desired Destination parameter(s), see "The Modulation Bus section".



Panel reference

Sound Engines On/Off and Edit Focus section



Engine Select

- **Click the On LED buttons to activate the corresponding Sound Engine.**
- **Click the I, II or III LED radio buttons to select the corresponding Sound Engine for editing.**

The Oscillator section



Here is where you choose oscillator waveform and set the wave shape and pitch for the selected Sound Engine.

On

- **Click the red rectangular LED button to switch the selected Sound Engine on/off.**

Oct

- **Set the pitch in octave steps.**
 Range: 5 octaves.

Semi

- **Set the pitch in semitone steps.**
 Range: 12 semitones (one octave).

Tune

- **Change the pitch in steps of 1 cent.**
 Range: +/- 50 cents (down or up half a semitone).

Kbd

→ **Set how much the pitch should track incoming MIDI Notes.**
Range: 0% (no tracking (constant pitch)) to 100% (1 semitone per note).

Waveform display

The interactive Waveform display shows the waveform shape in real-time.

- **Clicking and dragging vertically in the display changes the Shape parameter, see "Shape".**
- **Clicking and dragging horizontally in the display changes the Modifier 1 Amount parameter, see "Amount".**

► **See "Recording display movements in the sequencer" for tips about automating display movements.**

Waveform selector

→ **Click the Waveform name box to bring up a menu of the available waveforms.**
The wave shapes are shown in the display above and are updated in real-time according to the current settings and modulations. A great way to understand how the sound actually "looks".

The waveforms are:

- **Basic Analog**
A pure sinewave at Shape=0%, gradually transformed via triangle and square towards a sawtooth wave at Shape=100%.
- **Square-Ramp**
A square wave at Shape=0%, gradually transformed towards a sawtooth wave at Shape=100%.
- **Saw-Triangle**
A negative ramp sawtooth wave at Shape=0%, gradually transformed via triangle towards a positive ramp sawtooth wave at Shape=100%.
- **Pulse Width**
A 0% duty cycle pulse wave (silence) at Shape=0 gradually transformed via a 50% duty cycle square wave towards a 100% duty cycle pule wave (silence) at Shape=100%.

► **Modulate the Shape parameter from an LFO to achieve PWM, see "Shape Modulation" below.**

- **Game**
A lo-fi "early computer game" type of signal. Turn the Shape knob to change the overtone contents and the octave transposition.
- **Synced Sine**
A pure sinewave at Shape=0%. As the Shape is increased, the pitch of the synced sinewave oscillator is raised.
- **Formant Sweep**
A cosine window modulated by a sinewave. Turn the Shape knob to change the sinewave frequency and thus sweep through the generated formants.
- **Electro Mechanical**
This is a simulation of an electric piano. A soft/mellow tone at Shape=0% gradually transformed towards an agitated signal at Shape=100%, with natural sound at the 12 o'clock position (50%).
- **Vocal Cord**
A simulation of a vocal cord with a bit of noise modulation. Change the overtone content with the Shape knob.

► **Try this together with the Vocal Formants algorithm in the Spectral Filter section to generate "vocal" sounds, see "Vocal Formant".**

- **Karplus-Strong**
A physical model of a "string", generated by sending a short pulse through pitched delay lines. At Shape=0% there is no damping and at Shape=100% there is full damping, which results in just a short clicking sound.

► **Try this together with the "Stretch" algorithm in the Harmonics section to create realistic metallic sounds.**



- **Envelope 3-4**
 This is a special mode where you can manually draw your waveforms in the Envelope 3 and Envelope 4 windows and then gradually crossfade between the drawn waveforms using the Shape knob. See "Using the Envelope 3 and Envelope 4 curves as Sound Engine waveforms" for information on how to draw your own waveforms.
- **FM > FM Ratio (1:1, 1:2, 1:8, 2:1)**
 These are frequency modulated sine waves with different frequency ratios between the carrier (C:) and modulator (:M) signals. Set the frequency modulation amount with the Shape knob.
- **FM > FM Feedback**
 A pure sinewave signal at Shape=0% gradually fed back internally at an 1:1 ratio. The feedback signal is filtered before fed back to the carrier signal. If you modulate the Shape parameter from e.g. an LFO you will get a similar result as when using the FM FB Noise waveform without Shape modulation, see "Noise > FM FB Noise" below.
- **Noise > S/H Noise**
 A sample & hold modulated noise. Change the sample & hold rate with the Shape knob. If you play high up on the keyboard at high Shape values, you get a kind of "pitched noise" sound.

 ► **To get white noise, set Shape to max, set Oct to -1 and turn Kbd to 0.**
- **Noise > Perlin Noise**
 A pure sinewave signal modulated by low frequency noise. At Shape=0% the noise has its lowest frequency and at Shape=100% the noise frequency is higher (but still low-frequent). The character of the signal is similar to the Band Noise in the Thor synthesizer.
- **Noise > Bit Noise**
 This generates a random lo-fi "digital" bit noise. At Shape=0% the signal is completely silent and with increasing Shape values the signal is modulated faster and in a wider frequency range.
- **Noise > FM FB Noise**
 A pure sinewave signal at Shape=0% gradually fed back internally at an 1:1 ratio. The feedback signal unfiltered before fed back to the carrier signal which gives the signal a noisy character.

 ► **To get a cleaner FM signal, use the FM Feedback waveform, see "FM > FM Feedback" above.**
- **Noise > Freeze Noise**
 This signal produces a range of noises, from tonal noise up to almost white noise, by amplitude modulating the signal's partials with noise.
- **Wave Tables >**
 The Wave Tables sub-menu contains a selection of very useful wave tables. Each wave table features eight waveforms that you could crossfade between with the Shape knob.
- **User Wave/User Wave Smooth**
 The User Wave options let you use the external sample you have loaded in the User Wave section (see "The User Wave and Mixer section"). The oscillator then generates and plays back wavetables (grains) of that sample. The "User Wave Smooth" algorithm uses a crossfaded loop within each grain, which produces a smoother character to the sound. Set the playback position in the sample with the Shape knob. Modulate the Shape parameter, for example from a negative Envelope ramp, for continuous movement in the sample.

Shape

→ **Turn the Shape knob to change the shape of the currently selected waveform.**
The wave shapes are shown in the display above and are updated in real-time according to the current Shape settings.

Shape Modulation

→ **Click the Shape Modulation Source box to bring up a menu of the available modulation sources.**
The "Inverted" sub-menu contains inverted variations of all modulation sources.

→ **Set the modulation amount with the Shape Modulation Amount knob.**



→ **Turn the Velo knob to control the Shape Modulation Amount from Keyboard Velocity.**

► **If you want other modulation sources or scaling options, use the Mod Bus, see "The Modulation Bus section".**

Phase Sync

→ **Click the Phase Sync button to force the waveform cycle to always start at the same phase (0 degrees).**
When active, the sound character will be the same each time you play the same note. When inactive, the sound character will vary more or less each time you play the same note.

The Modifiers section



The two Modifiers can be used for modifying the currently selected waveform in various ways. The two Modifiers are identical in functionality and can be used alone or together (or not at all).

Modifier On/Off

→ **Click the On/Off LED buttons to activate/deactivate the corresponding Modifier.**

Modifier selector

→ **Click the Modifier name box to bring up a menu of the available Modifier types.**
The available Modifier types are:

- **Faded Sync**
 This is oscillator sync but with a crossfade at the sync positions. This makes the effect a little smoother (less bright) than with regular hard sync, see below.
- **Hard Sync**
 Oscillator hard sync is when one oscillator restarts the period of another oscillator, so that they will have the same base frequency. If you change or modulate the frequency of the synced oscillator you get the characteristic sound associated with oscillator sync. Control the frequency of the synced oscillator (and thereby the overtone spectrum) with the Amount knob.
- **Invert**
 This inverts the waveform phase at a variable position within the waveform cycle. Set the phase angle with the Amount knob.
- **Mirror**
 This mirrors the waveform cycle (in the time line) at a variable position in the waveform cycle. Set the mirroring position in the waveform cycle with the Amount knob. At Amount=50% the waveform is completely symmetric.
- **DownSample**
 This lets you quantize the waveform in time, i.e. reduce the sample rate. Set the sample rate reduction amount with the Amount knob.
- **Quantize**
 This lets you truncate the signal's bit depth, thus making it possible to achieve that noisy, characteristic "8-bit sound" for example. Set the bit-reduction amount with the Amount knob.

- **Phase Distort**
 This distorts the waveform by modulating the start phase of the waveform cycle. This generally creates a brighter tone towards the extremes of the Amount range (0% and 100%). At Amount=50% the signal is unaffected.
- **Self Multiply**
 This multiplies a copy of the waveform with the original waveform. Set the phase angle of the copied waveform with the Amount knob.
- **Noise Mod**
 This modulates the waveform with low frequency noise. Perfect for adding e.g. "breath noise" to a signal. Set the noise modulation amount with the Amount knob.
- **Shaping > Wrap**
 This amplifies the signal above the available headroom and then wraps the peaks down into the available headroom. This adds quite an aggressive distortion to the sound.
- **Shaping > Fold**
 This amplifies the signal above the available headroom and then "mirrors" the peaks down into the available headroom. Fold is similar to the Wrap shaping but is generally less "aggressive".
- **Shaping > Hard Clip**
 This amplifies the signal above the available headroom and then clips the peaks that are above the headroom. Generally, a signal that is clipped to the maximum would result in a pulse/square shaped waveform.
- **Shaping > Soft Clip**
 Soft clip is similar to hard clip described above, but has a smoother shape at the clipping points and thus generates less overtones.
- **Shaping > Sine Shape**
 This generates sine shaping distortion to the signal.
- **Shaping > Glitch 1**
 This distorts the waveform by introducing a short low-frequency noise glitch in the waveform cycle, but only in parts of the waveform that go from zero to positive level.
- **Shaping > Glitch 2**
 This is similar to Glitch 1 described above, but introduces a more high-frequent noise glitch in the waveform cycle.
- **Harmonics > Octave**
 This makes it possible to gradually crossfade between the original signal and a copy of the signal one octave above. Set the crossfade amount with the Amount knob.
- **Harmonics > Fifth**
 This makes it possible to gradually crossfade between "one octave up" and "one octave+one fifth up". As soon as you turn on the Fifth modifier you automatically raise the pitch by one octave. The reason for this is that this is done by multiplying frequencies, i.e. you crossfade between the double and triple of the original frequency. Set the crossfade amount with the Amount knob.
- **Harmonics > 16 Harmonics**
 This makes it possible to gradually crossfade between the original signal and copies of the signal at the 16 first harmonics above the original frequency. Set the position in the harmonic spectrum with the Amount knob.
- **Harmonics > Fund. + 16 Harm.**
 This is the same as "16 Harmonics" described above, except this always keeps the original signal mixed in with the selected harmonic.
- **Harmonics > All 16 Harm.**
 This gradually adds copies of the original signal at the first 16 harmonics. Turning up the Amount knob will add on the harmonics one by one until all 16 harmonics are present in the signal.
- **Harmonics > Ring Harm.**
 This multiplies the waveform with a sinewave signal to generate a ring modulator effect. Set the modulator frequency with the Amount knob.



- **FM > FM Ratio (1:1, 1:2, 1:8, 2:1)**
 These modifiers let you frequency modulate the currently selected Waveform at various ratios. The carrier signal is the currently selected Waveform (C:) and the modulator (:M) is the modifier signal. Set the frequency modulation amount with the Amount knob.
- **FM > FM Feedback**
 Here, an internally fed back sinewave signal at an 1:1 ratio modulates the waveform (same signal type as in the "FM > FM Feedback" Waveform).
- **Detuning > Unison3**
 This simulates 2 copies of the original signal. The Amount knob controls the detuning amount and rate.
- **Detuning > Unison7**
 This simulates 6 copies of the original signal. The Amount knob controls the detuning amount and rate.
- **Detuning > Ensemble**
 This simulates a variable number of copies of the original signal. The Amount knob controls the number of copies, the detuning amount and rate.
- **Detuning > Unison 3Oct**
 This simulates 2 copies of the original signal at +1 and +2 octaves relative to the original signal. The Amount knob controls the detuning amount and rate.
- **Formant**
 This simulates a formant (body) filter, which produces multiple peaks and notches in the frequency spectrum of the signal. The Amount knob controls the formant transposition in the frequency spectrum. At Amount=50% the signal is unaffected. Below 50% the formant is transposed down and above 50% it's transposed up.

 To make the formant static in the frequency spectrum, regardless of which note you play, modulate the Amount parameter using the "-KEY" (inverted) modulation source with a fairly high value (see "Amount Modulation" below). This is especially useful if you are using an acoustic instrument sample as User Wave in the Oscillator section.

Amount

→ **Turn the Amount knob to change the modification amount of the currently selected Modifier.**
The wave shapes are updated in real-time and shown in the Waveform display.

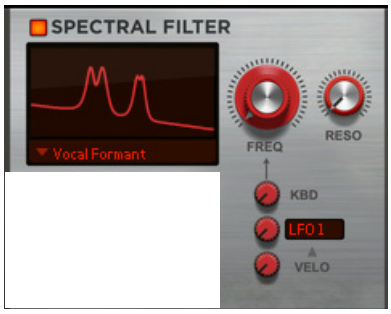
Amount Modulation

→ **Click the Modulation Source box to bring up a menu of the available modulation sources.**
The "Inverted" sub-menu contains inverted variations of all modulation sources.

→ **Set the modulation amount with the Amount Modulation knob.**

► **If you want other modulation sources or scaling options, use the Mod Bus, see "The Modulation Bus section".**

The Spectral Filter



The signal from the Oscillator section can then be processed by the Spectral Filter. The Spectral Filter features a wide variety of algorithms that affect the partials of the signal.

Spectral Filter On/Off

→ **Click the On/Off LED button to activate/deactivate the Spectral Filter.**

Spectral Filter display

The interactive Spectral Filter display shows the filter shape in real-time.

- **Clicking and dragging vertically in the display changes the Freq parameter, see "Freq".**
- **Clicking and dragging horizontally in the display changes the Resonance parameter, see "Reso".**

← **See "Recording display movements in the sequencer" for tips about automating display movements.**

Spectral Filter selector

→ **Click the Spectral Filter name box to bring up a menu of the available filter types.**
The available filter types are:

- **LP 12**
 This simulates a standard 12dB/octave lowpass filter. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob.
- **LP 24**
 This simulates a standard 24dB/octave lowpass filter. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob.
- **HP 24**
 This simulates a standard 24dB/octave highpass filter. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob.
- **BP 12**
 This simulates a standard 12dB/octave bandpass filter. Set the center frequency with the Freq knob and the resonance amount with the Reso knob.
- **Par EQ**
 This simulates a standard single band parametric equalizer with a fixed bandwidth. Set the center frequency with the Freq knob and the gain/attenuation with the Reso knob.
- **Dual Peak**
 This simulates two 12dB/octave bandpass filter routed in parallel. Set the center frequency of the first bandpass filter with the Freq knob and the peak separation with the Reso knob.
- **Vocal Formant**
 This simulates the formants of the vocal tract by using multi-peak+notch filters. Change the formant with the Freq and Reso knobs.
- **LP Variable Slope**
 This simulates a non-resonant lowpass filter with a variable attenuation slope. Set the cutoff frequency with the Freq knob and the attenuation slope with the Reso knob.
- **HP Variable Slope**
 This simulates a non-resonant highpass filter with a variable attenuation slope. Set the cutoff frequency with the Freq knob and the attenuation slope with the Reso knob.
- **Comb +**
 This simulates a multi notch filter, great for phaser types of effects. Set the cutoff frequency of the first notch with the Freq knob and the attenuation amount - and consequently the bandwidth - of the notches with the Reso knob. The difference between "Comb +" and "Comb –" (see below) is in the position of the peaks in the spectrum. The main audible difference is that the "Comb –" version causes a bass cut.



- **Comb -**
 This simulates a comb filter with a positive feedback loop - but without feed forward - ideal for flanger and phaser types of effects. Set the cutoff frequency of the second peak with the Freq knob and the resonance amount with the Reso knob. The difference between "Comb +" (see above) and "Comb –" is in the position of the peaks in the spectrum. The main audible difference is that the "Comb –" version causes a bass cut.
- **Resonator 1,2 and 3**
 The three Resonator algorithms contain formant filter tables that simulate various body resonances (multi-peak+notch filters). Set the position in the formant tables with the Freq knob and the resonance with the Reso knob.
- **Envelope 4**
 This is a special mode where you can manually draw your own filter curve in the Envelope 4 window. You then control the cutoff/center frequency with the Freq knob and the resonance with the Reso knob. See "Using the Envelope 4 curve as a Spectral Filter curve" for information on how to draw your own filter curves.
- **User Wave**
 This utilizes a filter generated from FFT analyses of the external sample you have loaded in the User Wave section (see "The User Wave and Mixer section"). Transpose the formant up/down in the frequency spectrum with the Freq knob and change the filter's position in the sample with the Reso knob.

► **To create a classic "vocoder" effect, load a vocal/speech sample in the User Wave section. Then, set the Freq knob to 50%, the Reso knob to 0% and the KBD knob to 0%. Then, have one of the Envelopes modulate the Reso parameter using the Modulation Bus (see "The Modulation Bus section"). Create a positive ramp (inverted sawtooth) envelope and set the Reso modulation amount to 100%.**

Freq

→ **Set the cutoff/center frequency of the currently selected Spectral Filter type.**

Reso

→ **Set the resonance amount of the currently selected Spectral Filter type.**

Frequency Modulation

→ **Turn the Kbd knob to set the keyboard tracking amount.**
At 0% the filter is static and doesn't track the keyboard at all. At 100% the filter tracks the keyboard 1 semitone per note. At values above 0% you can also see the filter curve move sideways in the Spectral Filter Display depending on where on the keyboard you play.

→ **Click the Frequency Modulation Source box to bring up a menu of the available modulation sources.**
The "Inverted" sub-menu contains inverted variations of all modulation sources.

→ **Set the modulation amount with the Frequency Modulation Amount knob.**

→ **Turn the Velo knob to control the Frequency Modulation Amount from Keyboard Velocity.**

► **If you want other modulation sources or scaling options, use the Mod Bus, see "The Modulation Bus section".**

The Harmonics section



The Harmonics section offers extensive modulation possibilities of the partials of the signal. For most algorithms the partials' characteristics is displayed in the Spectral Filter display, see "Spectral Filter display".

Harmonics On/Off

→ **Click the On/Off LED buttons to activate/deactivate the Harmonics section.**

Harmonics selector

→ **Click the Harmonics name box to bring up a menu of the available harmonic algorithms.**
The available Harmonics types are:

- **Random Gain**
 This alters the gain for each of the partials in the signal in a random fashion. Turn the Pos knob to change the randomization "pattern" and the Amount knob to change the partial gain levels in the "pattern".
- **Harmonic 1-8 Mix**
 This alters the gain/attenuation for the first eight partials in the signal. Turn the Pos knob to crossfade between the partials and the Amount knob to change the partial gain/attenuation level. Amount levels below 50% attenuate all partials but the one selected with the Pos knob. Amount levels above 50% attenuate the partial selected with the Pos knob.
- **Odd-Even**
 This alters the gain/attenuation of the partials in the signal. At Pos=0% the Amount knob controls the mix strictly between the odd and even partials in the signal. At other Pos values, the gain/attenuation is not strictly on odd and even partials. At Amount=50% the Pos value has no effect.
- **Stretch**
 This stretches or squeezes all partials (overtones) - except for the fundamental - in the signal, up or down in the frequency range. Perfect for turning a harmonic signal into a more inharmonic one. Change the stretch amount with the Amount knob. The Pos knob controls the start phase of all the overtones. At Pos=0% all partials start at the same phase. When Amount is set fairly high the Pos parameter have little or no influence on the sound.

 ► **Try this with the Karplus-Strong waveform to create really realistic metallic sounds, see "Karplus-Strong"!**
- **Ensemble**
 This is the perfect algorithm for really dense pad sounds. The Ensemble algorithm simulates a type of chorus effect by utilizing noise modulation of the partials. Set the noise frequency with the Pos knob and the mix level with the Amount knob.
- **Ensemble Sparse**
 The Ensemble Sparse algorithm also utilizes noise modulation of the partials, but here a lot of noise frequency bands are cut out. This makes Ensemble Sparse sound more animated and less smooth than the Ensemble algorithm described above. Set the noise frequency with the Pos knob and the mix level with the Amount knob.
- **HF Noise**
 This amplitude modulates the high frequencies in the signal with (high-frequency) noise, perfect for adding "breath noise" to the signal, for example. Set the noise frequency with the Pos knob and the noise mix level with the Amount knob.
- **Harmonic Lag A-R**
 The Harmonic Lag A-R algorithm is designed especially for use with the User Wave algorithm in the Spectral Filter (see "User Wave") to create vocoder effects. The Harmonic Lag A-R algorithm controls the Spectral Filter - so the Spectral Filter has to be on for this to work!

 Note that the Harmonic Lag A-R algorithm works on the filter partials - not the oscillator's signal partials. Set the Attack time of the filter partials with the Pos knob and the Release time with the Amount knob. These controls work similarly to the Attack and Decay parameters on the BV512 Vocoder device.

Pos

→ **Turn the Pos knob to change the frequency of the currently selected Harmonics algorithm.**
The frequency spectrum is updated in real-time and shown in the Spectral Filter display.



Amount

→ **Turn the Amount knob to change the intensity of the currently selected Harmonics algorithm.**
The frequency spectrum is updated in real-time and shown in the Spectral Filter display.

The Unison section



The Unison function generates detuned duplicates of the signal in pairs on either side of the original signal's pitch.

Unison On/Off

→ **Click the On/Off LED button to activate/deactivate the Unison section.**

Unison display

The Unison display shows the unison characteristics, as set with the controls in the Unison section. Note, though, that this display is not interactive like the Waveform and Spectral Filter displays.

Unison Type selector

→ **Click the Unison name box to bring up a menu of the available Unison types.**
The available Unison types are:

- **Normal**
 This generates duplicates of the signal on either side of the original signal's pitch.
- **Fourth**
 This generates duplicates of the signal on either side of the fourth above the original signal's pitch.
- **Fifth**
 This generates duplicates of the signal on either side of the fifth above the original signal's pitch.
- **Octave Down**
 This generates duplicates of the signal on either side of the original signal's pitch - one octave down.
- **Phase Only**
 This generates duplicates of the signal on either side of the original signal's pitch. The Detune parameter (see below) now controls the phases of the signal copies, instead of the detuning. This is great for creating wide stereo sounds without lots of detuning.

 If Phase Sync is active in the Oscillator (see "Phase Sync"), all signals in the Unison function get the same relative start phases each time you play the same note. The original signal always has the phase 0 degrees.

Count

→ **Set the number of desired signals in the unison effect.**
Range: 1-7.
Note that for even numbers, the original signal is represented by two duplicates.

Blend

→ **Set the mix between the original signal and the duplicates.**
For even "Count" numbers (see above), one of the duplicates represents the original signal in the mix.

Detune

→ **Set the pitch detuning of the signal duplicates.**
If the "Phase Only" Unison type is selected (see above), the Detune knob controls the phases of the signal duplicates instead of the pitch detuning.

► **In the "Phase Only" scenario it could also be a good idea to modulate the Detune parameter from an LFO using the Modulation Bus (see "The Modulation Bus section"), to create nice phasing effects.**

Spread

→ **Set the stereo spread amount of the signal duplicates.**

The User Wave and Mixer section



The User Wave and Mixer section is where you can load an external sample to use in the Oscillators (see "User Wave/User Wave Smooth") and/or in the Spectral Filter (see "User Wave"). In the Mixer you can mix and pan the signals from the three Sound Engines before sending them to the Filter, Amp and Multi FX sections.

Sample Select/Load/Edit buttons

→ **Load a sample using drag & and drop, or by clicking the Browse sample button, or by using the Up/Down buttons to scroll and load a sample from the currently selected folder.**

• **It's possible to load stereo samples. However, the sample will be automatically converted to mono in Europa.**

! **Like with the other "sampler" devices in Reason, the Europa patch does not include the actual sample - only a reference to it. Therefore, the sample has to be stored separately on disk or in a ReFill on your computer.**

! **If the sample length is a multiple of 2048 samples ("Serum compatible"), no pitch detection is being made. Then, Europa automatically assumes that 2048 samples is one complete waveform cycle (period). If the sample is not an exact multiple of 2048 samples, a pitch detection is being performed by Europa to determine the pitch. Longer samples (with a stable pitch) will render better pitch detection results, so don't use very short samples.**

Level

→ **Set the volume of the corresponding Sound Engine signal with the Level slider.**
Range: -Inf to +6.0dB.

Pan

→ **Set the panning of the corresponding Sound Engine signal in the stereo panorama.**

The Filter section



Routing buttons

→ **Click the red LED buttons to route the corresponding Sound Engine signals to the Filter section.**
If deactivated, the signal bypasses the Filter and goes straight to the Amp section, see "The Amplifier section".

Drive

→ **Turn the Drive knob to amplify and introduce an overdrive type of distortion to the Sound Engine signal(s) in the filter.**

Filter Type selector

→ **Click the Filter Type name box to select one of the following filter types:**

- **SVF HP 12dB**
 A state variable (SVF) highpass filter with a 12dB/octave slope. This filter is similar to the State Variable Filter in the Thor synthesizer.
- **SVF BP 12dB**
 A state variable (SVF) bandpass filter with 12dB/octave slopes. This filter is similar to the State Variable Filter in the Thor synthesizer.
- **SVF LP 12dB**
 A state variable (SVF) lowpass filter with a 12dB/octave slope. This filter is similar to the State Variable Filter in the Thor synthesizer.
- **SVF Notch**
 A state variable (SVF) notch filter. This filter is similar to the State Variable Filter in the Thor synthesizer.
- **Ladder LP 24dB**
 A ladder-type lowpass filter with a 24dB/octave slope. The resonance peak more narrow in this filter type than in the MFB LP 24dB filter (see below). The filter can be driven to self-oscillate.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

- **MFB LP 12dB**
 A multiple feedback (MFB) lowpass filter with a 12dB/octave slope.
- **MFB LP 24dB**
 A multiple feedback (MFB) lowpass filter with a 24dB/octave slope. The resonance peak is wider in this filter type that in the Ladder filter (see above). The filter can be driven to self-oscillate.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

- **MFB HP 24dB**
 A multiple feedback (MFB) highpass filter with a 24dB/octave slope.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

- **K35 LP 12dB**
 An "early MS-20 type" of lowpass filter with a 12dB/octave slope. The filter can be driven to self-oscillate.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

Reso

→ **Set the resonance amount.**
In the SVF Notch filter, the Reso knob controls the width of the notch - from wide to narrow.

Freq

→ **Set the cutoff frequency (for the HP and LP filter types) or the center frequency (for the BP filter type).**

Frequency Modulation

→ **Turn the Kbd knob to set the keyboard tracking amount.**
At 0% the filter is static and doesn't track the keyboard at all. At 100% the filter tracks the keyboard 1 semitone per note.

→ **Click the Frequency Modulation Source box to bring up a menu of the available modulation sources.**
The "Inverted" sub-menu contains inverted variations of all modulation sources.

► **Use one of the Envelopes (see "The Envelopes section") as modulation source to create a Filter envelope.**

→ **Set the modulation amount with the Frequency Modulation Amount knob.**

→ **Turn the Velo knob to control the Frequency Modulation Amount from Keyboard Velocity.**

► **If you want other modulation sources or scaling options, use the Mod Bus, see "The Modulation Bus section".**



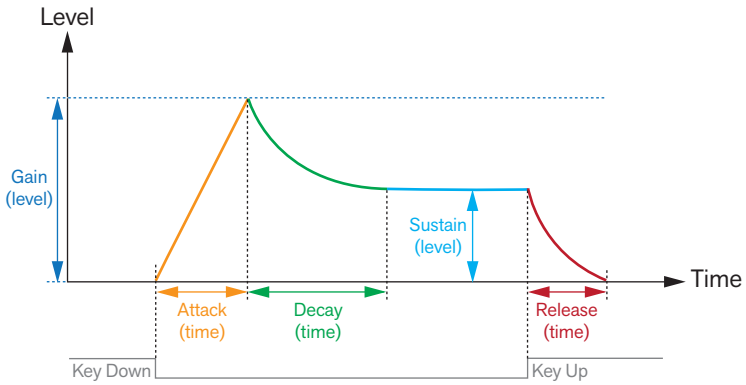
The Amplifier section



The Amplifier section contains a standard ADSR envelope, which controls the amplitude of the signals from all three Sound Engines equally.

► **To create an "amp envelope" for a separate Sound Engine, have a look at "Creating an individual "pre amp envelope" for a Sound Engine".**

The picture below shows the various stages of the ADSR envelope:



The ADSR envelope stages.

A(ttack)

When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the value set with the Gain knob (see below). How long this should take, depends on the Attack setting. If the Attack is set to "0", the Gain level is reached instantly. If the Attack value is raised, it will take longer time before the Gain level is reached.

D(ecay)

After the Gain level has been reached, the level starts to drop. How long this should take is governed by the Decay parameter.

If you want to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0", the Decay parameter should be set to a medium value and the Sustain level should be set to "0", so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you raise the Sustain parameter.

S(ustain)

The Sustain parameter determines the level the envelope should rest at, after the Decay stage. If you set Sustain to full level, the Decay setting is of no importance since the volume of the sound is never lowered.

If you want to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

But often a combination of Decay and Sustain is used to generate envelopes that rise up to the Gain level, then gradually decreases to finally land to rest on a level somewhere in-between zero and the Gain level. Note that Sustain represents a level, whereas the other envelope parameters represent times.

R(elease)

The Release parameter works just like the Decay parameter, except it determines the time it takes for the volume to drop back to zero after you release the key.

Pan

- **Set the panning of the output signal from the Amplifier in the stereo panorama.**
- **Since Pan works individually per voice, you can assign e.g. Keyboard Velocity or an Envelope in the Modulation Bus to control the Pan effect, see "The Modulation Bus section".**

Gain

- **Set the desired maximum level for the Amplifier with the Gain knob.**
 This is the maximum level the envelope will reach after the Attack stage is completed (see above).
- **If you want to create a tremolo effect, assign "Gain" as Destination and an LFO as Source in the Modulation Bus section, see "The Modulation Bus section".**

Velo

- **If you want the Gain level to be controlled from keyboard velocity, turn up the Velo knob.**



The Envelopes section



The Envelopes section features four separate polyphonic (one per voice) general purpose envelope generators, that can be assigned to control selectable parameter(s) in the Modulation Bus section.

The Envelopes are extremely flexible, and you can draw your own custom modulation shapes by clicking and drawing in the display area. There are also a number of preset shapes that you can use as starting points (or use as is). If you use Loop mode, you could turn the envelope into a kind of LFO.

See "The Modulation Bus section" for details on how to assign the Envelopes to the desired destination(s).

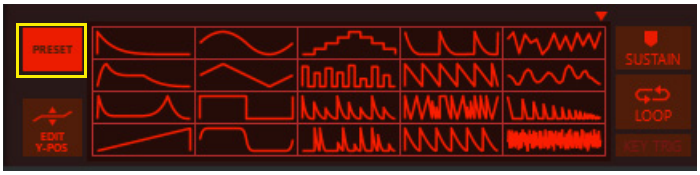
Envelope 1, 2, 3 and 4

→ **Click one of the Envelope 1, 2, 3 or 4 buttons to select which envelope to edit:**

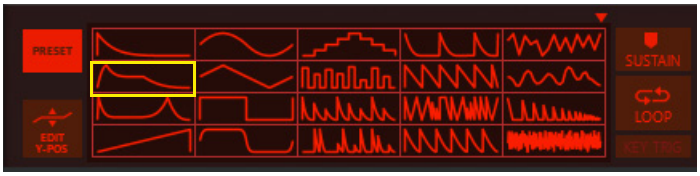


Preset

1. **Click the Preset button to bring up a palette of envelope preset curves:**



2. **Click the desired envelope preset curve to place it on the display.**
 Let's select a standard ADSR style of envelope curve:





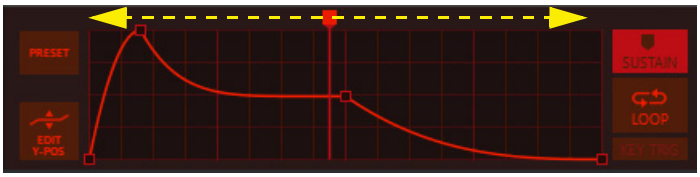
Adding a Sustain stage

→ **Click the Sustain button to add a sustain stage to the envelope:**



The vertical red marker that appears indicates at what level (and where) the envelope will stay sustained until you release the key.

→ **Drag the sustain marker sideways to move the sustain stage to the desired position:**



→ **To remove the sustain stage, click the Sustain button.**

Adding and removing envelope points

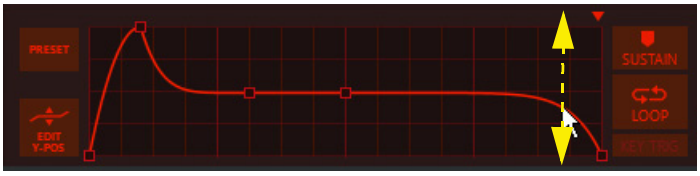
→ **Double click, or hold down [Ctrl](win) or [Cmd](Mac) and click in the envelope display to add points to the envelope curve:**



→ **To remove a point, double click, or hold down [Ctrl](win) or [Cmd](Mac) and click, on an existing point on the envelope curve.**

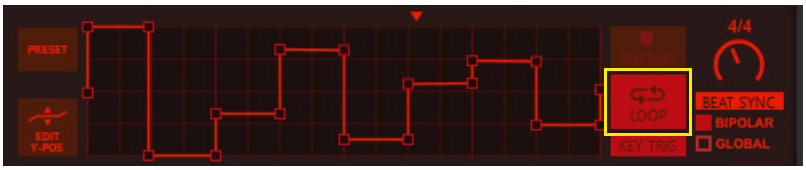
Changing the envelope curve shape

→ **Click a line segment (between two points) and drag up/down to change the curve shape:**



Looping the envelope

→ **Click the Loop button to turn the envelope into a kind of LFO.**
If there was previously a sustain stage in the envelope, this will automatically be disabled when you click the Loop button.

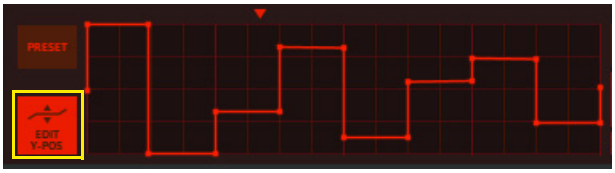


Here we have edited a stepped curve from the Presets. We have also enabled Beat Sync and set the length/rate to 4/4. This means that each step in the curve now represents an 1/8th note.

- **Key Trig means the envelope restarts when you play a note.**
- **You can choose whether the envelope should send out a bipolar value or unipolar one (0-100%).**
- **If Global is on, the envelope will be common for all voices.**

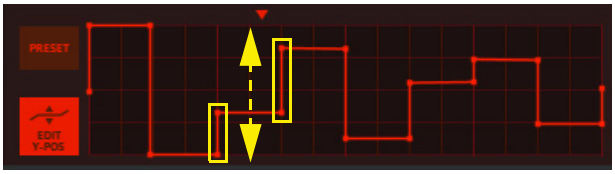
Editing levels only

→ **To restrict the editing to levels only, without affecting the time positions, click the Edit Y-Pos button:**



In this mode you cannot change the time positions of the envelope points, only their levels (height). This is extra useful with a stepped Preset curve, because dragging up or down will change the value of an entire segment, turning the Envelope into a pseudo-sequencer.

! **To be able to adjust the level of a segment, the two points on either side of the segment have to be on the exact same time positions. Otherwise, only the closest point will be changed. Also, any inclining/declining segment will automatically turn horizontal when edited:**

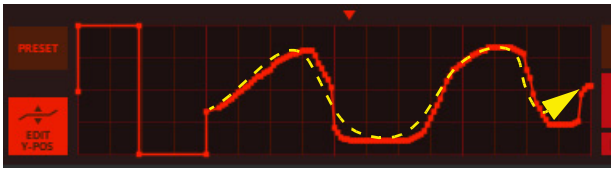


Adjusting the level of a segment.

Creating “free form” envelope curves

In the Edit Y-Pos mode, you can also draw “free form” curves:

→ **To continuously add new consecutive points, hold down [Ctrl](win) or [Cmd](Mac) and drag in the envelope display:**



→ **To erase points, hold down [Shift] and [Ctrl](win) or [Cmd](Mac) and drag in the envelope display.**

Using the Envelope 3 and Envelope 4 curves as Sound Engine waveforms

As a special feature you can use the Envelope 3 and Envelope 4 curves as waveforms for the Sound Engines:

1. **Select Envelope 3 and create/modify a curve in the envelope display:**



2. **Select Envelope 4 and create/modify another curve:**



3. **In the Waveform selector for a Sound Engine, select the "Envelope 3-4" waveform:**



4. **Turn the Shape knob to crossfade between the curves/waveforms of Envelopes 3 and 4:**



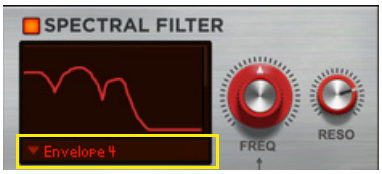
Using the Envelope 4 curve as a Spectral Filter curve

Another special feature is that you could use the Envelope 4 curve as a filter curve in the Spectral Filter:

1. **Select Envelope 4 and create/modify a curve in the envelope display:**



2. **In the Filter selector in the Spectral Filter section, select "Envelope 4":**



3. **Turn the Freq knob to change the curve's "cutoff" frequency and the Reso knob to change the curve's "resonance".**
 At Reso=0% the curve is completely flat (no gain or attenuation) and at Reso=100% the resonance corresponds exactly to the Envelope 4 curve.

The LFO section



An LFO (Low Frequency Oscillator) is used for generating cyclic modulation. A typical example is to have an LFO modulate the pitch of a signal to produce vibrato, but there are countless other applications for LFOs.

The LFO section features three separate general purpose LFOs, that can be assigned to control selectable parameter(s) in the Modulation Bus section, see "The Modulation Bus section".

1. **Select which of the three LFOs you want to edit by clicking one of the LFO 1, LFO 2 and LFO 3 buttons.**
2. **Select an LFO waveform by clicking the spin controls to the right of the waveform display, or by click-holding in the display and dragging up or down.**
 Besides the standard waveforms (sine, triangle, pulse, etc.) there are random, slope and stepped waveforms. The shape of the waveforms are shown in the display.
3. **Set the LFO frequency with the Rate knob.**

- **Click the Beat Sync button to sync the LFO to the main sequencer tempo.**
 The Rate parameter now controls the time divisions.
- **Click the Key Sync button to restart the LFO at every new Note On.**
- **Click the Global button to make the LFO common for all voices (monophonic).**
- **Turn the Delay knob to introduce a delay before the LFO modulation kicks in after a note is played.**
 Turn clockwise for longer delay times.

The Effects section



The Effects section features six different effect modules that can be freely reordered by dragging & dropping. Most of the effect parameters are also available as destinations in the Modulation Bus, see "The Modulation Bus section".

At the top of the Effects section are six Effect buttons. Click any of these to bring up the control panel for the corresponding effect. Below the Effect buttons are the On/Off buttons for the individual effects. Click these to activate the effects.

Reordering the effects

→ **To define the order of the effects in the serial chain, click and hold on the desired Effect button and drag sideways to the desired position:**



Moving the Reverb effect to another position in the effects chain.

You can reorder the effects at any time.

Reverb



This is a stereo reverb, routed as a send effect.

- **Decay**
 This governs the length of the reverb effect.
- **Size**
 Sets the emulated room size, from small room to large hall. Middle position is the default room size.
 Lowering this parameter results in a closer and gradually more "canned" sound. Raising the parameter results in a more spacey sound, with longer pre-delay.
- **Damp**
 Raising the Damp value cuts off the high frequencies of the reverb, thereby creating a smoother, warmer effect.
- **Amount**
 Use this parameter to adjust the send level to the Reverb effect.

 If you play a note, have a long delay Decay time and turn down Amount, the reverberation will continue.

Delay



This is a stereo delay, routed as a send effect.

- **Sync**
 Activate Sync to sync the delay time to the main sequencer tempo.
- **Time**
 This sets the time between the delay repeats. If Sync is active (see above), the Time parameter now controls the time divisions.
- **Ping Pong**
 Activate Ping Pong to have the delay repeats alternating between left and right in the stereo panorama. The effect is also dependent on the Pan parameter (see below).
- **Pan**
 Sets the panning of the delay repeats in the stereo panorama. If Ping Pong is active (see above) the Pan knob controls the panning of the initial delay repeat as well as the total stereo spread of the remaining repeats.
- **FB**
 The FB (feedback) parameter determines the number of delay repeats.
- **Amount**
 Use this parameter to adjust the send level to the Delay effect.

► **If you play a note, have a long delay feedback and turn down Amount, the echoes will continue. This allows for automated "triggered delay" fx.**

Distortion



The Distortion effect features six different types of distortion.

→ **Select distortion type with the switch.**
"Dist" produces a dense, rich analog type of distortion.
"Scream" produces a less bright type of distortion.
"Tube" emulates a tube type of distortion.
"Sine" is a sine shaping distortion.
"S/H" gives the effect of sample rate reduction.
"Ring" is a ring modulator effect.

- **Drive**
 Sets the overdrive/feedback level of the selected distortion.
- **Tone**
 This is a lowpass filter and sets the tone of the selected distortion.
- **Amount**
 Sets the Dry/Wet amount of the distortion.



Compressor



This is a stereo compressor.

- **Attack**
 This governs how quickly the compressor will apply its effect when signals rise above the set threshold. If you raise this value, the response will be slower, allowing more of the signal to pass through the compressor unaffected. Typically, this is used for preserving the attacks of the sounds.
- **Release**
 When the signal level drops below the set threshold, this determines how long it takes before the compressor lets the sound through unaffected. Set this to short values for intense, "pumping" compressor effects, or to longer values for a smoother change of the dynamics.
- **Thres**
 This is the threshold level above which the compression sets in. Signals with levels above the threshold will be affected, signals below it will not. In practice, this means that the lower the Threshold setting, the more the compression effect.
- **Ratio**
 This specifies the amount of gain reduction applied to the signals above the set threshold.

Phaser/Flanger/Chorus



This is a stereo Phaser/Flanger/Chorus.

→ **Select effect type with the Phaser/Flanger/Chorus switch.**
The selected effect type is displayed on the Effect button.

- **Depth**
 Sets the depth of the selected effect. To get a static sound, set Depth to zero.
- **Rate**
 Sets the rate/speed of the modulation.
- **Spread**
 Sets the stereo width of the effect.
- **Amount**
 Sets the Dry/Wet amount of the effect.

► **It's also possible to modulate the start/center frequency of the Phaser/Flanger/Chorus using the "Chorus/Flanger/Phaser > Frequency" destination in the Modulation Bus, see "The Modulation Bus section".**

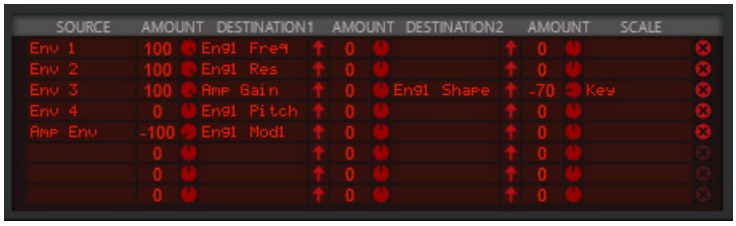
EQ



The EQ effect is a single band parametric equalizer with adjustable Q-value and Gain.

- **Freq**
 Sets the center frequency of the EQ band.
- **Q**
 Sets the bandwidth of the EQ band, from wide to narrow.
- **Gain**
 Sets the gain/attenuation of the EQ band, from -18dB to +18dB.

The Modulation Bus section



The Modulation Bus section is used for routing a modulation Source to one or two modulation Destinations each. This creates a very flexible routing system that complements the “pre-wired” routing in Europa.

The Modulation Bus section in Europa is derived from the one in the Reason Thor Polysonic Synthesizer device, so if you are familiar with Thor, you will quickly find your way around in Europa’s modulation bus.

There are eight “Source –> Destination 1 –> Destination 2 –> Scale” busses, of which the first four have pre-assigned sources. However, these four pre-assigned sources can be easily changed if you like.

A Source parameter can modulate two different Destination parameters per bus (with variable Amount settings). Each bus also has a Scale parameter that affects the relative modulation Amount for both Destinations.

► **Note that it is possible to assign the same source parameter as Source in several busses. This allows you to control more than two Destination parameters from the same Source.**

1. Select the desired Source parameter by clicking in the corresponding Source box and selecting from the list.

The following parameters can be used as modulation Sources:

Parameter	Description
Velocity	This applies modulation according to the Keyboard Velocity values (how hard or soft you strike the MIDI keyboard keys).
LFOs (LFO 1, LFO 2 and LFO 3)	This allows you to modulate parameters from LFO 1, LFO 2 and LFO 3 respectively.
Envelopes (Amp Envelope, Envelope 1, Envelope 2, Envelope 3, Envelope 4, Envelope 3 * Envelope 4, Envelope 3 * LFO 3)	This allows you to modulate parameters from any of the Envelopes. As a special feature you can also modulate parameters from the multiplied signal of Envelope 3 and Envelope 4, as well as from the multiplied signal of Envelope 3 and LFO 3.
Mod Wheel	This allows you to modulate parameters from the Mod Wheel.

Parameter	Description
MW Latched	This allows you to modulate parameters based on the current Mod Wheel value at a given Note On.
Pitch Wheel	This allows you to modulate parameters from the Pitch Bend control.
Breath	This allows you to modulate parameters from the Breath performance controller
Expression	This allows you to modulate parameters from the Expression performance controller
Aftertouch	This allows you to modulate parameters from Keyboard Aftertouch (channel aftertouch)
Sustain	This allows you to modulate parameters from a connected sustain pedal. Note that continuous sustain data (0-127) is supported - not just on/off.
Key	This is keyboard tracking. If a positive Amount value is used and the destination is filter frequency, the filter frequency will track the keyboard, i.e. increase with higher notes.
Random	This sends out a random value each time a new note is played.
Key In Octave	This allows you to modulate parameters based on 12 separate note values (within each octave).
Noise	This allows you to modulate parameters from white noise.
Polyphony	This allows you to modulate parameters based on the number of playing voices at a given time.
Last Velocity	This applies modulation according to the latest Keyboard Velocity value (how hard or soft you hit the latest MIDI keyboard key).
CV Input 1/2/3/4	This takes the current value on the CV 1/CV 2/CV 3/CV 4 inputs on the rear panel and sends to the desired destination.
CV Input 1/2/3/4 Latched	This allows you to modulate parameters based on the current CV 1/CV 2/CV 3/CV 4 value at a given Note On.

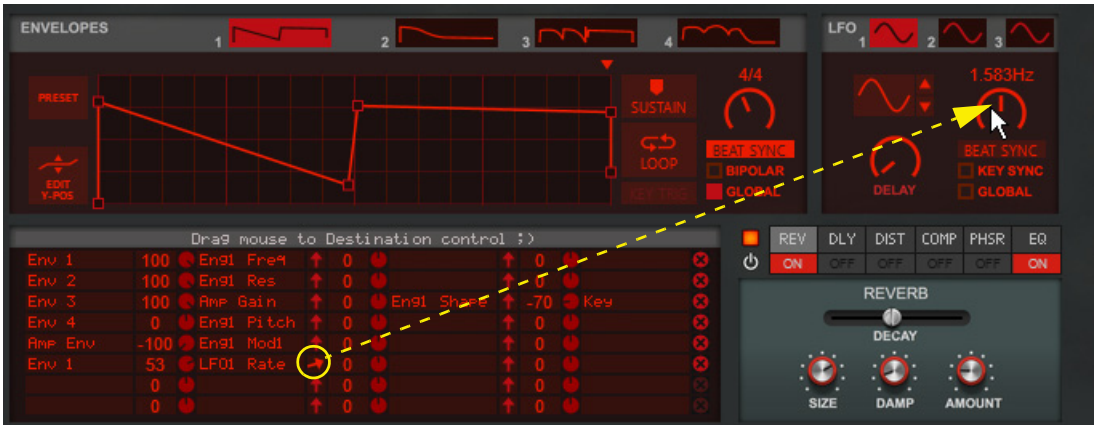
Modulation Bus Source parameters.

2. **Set the Amount for the first Destination by turning the corresponding Amount knob, or by clicking and dragging vertically in the corresponding Amount box.**

! **Note that the Amount range is +/-100. This means that the Amount value can exceed the modulated parameter's range. When this happens, the modulated parameter simply stays at its extreme value until the Amount value gets within the parameter's range again.**

3. **Select the first Destination parameter by click-holding the red arrow symbol to the right of the corresponding Destination box.**

4. **While click-holding, drag to the desired destination parameter on the panel:**



Assigning LFO 1 Rate as a Destination for Envelope 1.

As you hover over a valid destination control on the panel, the parameter name is automatically displayed in the Destination box in the Modulation Bus.

5. **To assign the currently selected Destination control, release the mouse button.**

→ **Alternatively, click the desired Destination box and select the Destination parameter from the list.**

The following parameters can be used as modulation Destinations:

Parameter	Description
Engine: Pitch	This affects the (full range) pitch of the Oscillator.
Engine: Shape	This affects the Shape parameter in the Oscillator section.
Engine: Mod 1 Amount	This affects the Modifier 1 Amount parameter in the Sound Engine.
Engine: Mod 2 Amount	This affects the Modifier 2 Amount parameter in the Sound Engine.
Engine: Filter Freq	This affects the Spectral Filter Frequency parameter in the Sound Engine.
Engine: Filter Res	This affects the Spectral Filter Resonance parameter in the Sound Engine.
Engine: Harmonics Pos	This affects the Harmonics Pos parameter in the Sound Engine.
Engine: Harmonics Amount	This affects the Harmonics Amount parameter in the Sound Engine.
Engine: Unison Count	This affects the Unison Count parameter in the Sound Engine.
Engine: Unison Detune	This affects the Unison Detune parameter in the Sound Engine.
Engine: Unison Blend	This affects the Unison Blend parameter in the Sound Engine.
Engine: Unison Spread	This affects the Unison Spread parameter in the Sound Engine.
Mixer: Level	This affects the Sound Engine Level in the Mixer section.
Mixer: Pan	This affects the Sound Engine Pan in the Mixer section
Filter: Drive	This affects the Drive parameter in the Filter section.
Filter: Freq	This affects the Frequency parameter in the Filter section.
Filter: Reso	This affects the Resonance parameter in the Filter section.
Amplifier: Gain	This affects the Gain parameter of the Amplifier section.
Amplifier: Pan	This affects the Pan parameter of the Amplifier section.
Amp Envelope: Attack	This affects the Attack time of the Envelope in the Amplifier section.
Amp Envelope: Decay	This affects the Decay time of the Envelope in the Amplifier section.
Amp Envelope: Sustain	This affects the Sustain level of the Envelope in the Amplifier section.
Amp Envelope: Release	This affects the Release time of the Envelope in the Amplifier section.
LFOs: Delay	This affects the LFO Delay parameters.
LFOs: LFO Rate	This affects the LFO Rate parameters.
Envelopes: Env Rate	This affects the Envelope Rate parameters.
Portamento	This affects the Portamento Time parameter.
CV Outputs: CV1/2/3/4 Out	This sends out the source modulation value(s) on the CV1/2/3/4 Output on the rear panel.
Reverb: Decay	This affects the Decay parameter in the Reverb effect.
Reverb: Amount	This affects the Amount parameter in the Reverb effect.
Delay: Time	This affects the Time parameter in the Delay effect.
Delay: Feedback	This affects the FB parameter in the Delay effect.
Delay: Amount	This affects the Amount parameter in the Delay effect.
Delay: Pan	This affects the Pan parameter in the Delay effect.
Dist: Drive	This affects the Drive parameter in the Dist effect.
Dist: Tone	This affects the Tone parameter in the Dist effect.
Dist: Amount	This affects the Amount parameter in the Dist effect.
Compressor: Release	This affects the Release parameter in the Compressor effect.
Compressor: Ratio	This affects the Ratio parameter in the Compressor effect.
Chorus/Flanger/Phaser: Frequency	This affects the center frequency of the Chorus/Flanger/Phaser effects.
Chorus/Flanger/Phaser: Amount	This affects the Amount parameter of the Chorus/Flanger/Phaser effects.
Par EQ: Frequency	This affects the Freq parameter in the EQ effect.
Par EQ: Gain	This affects the Gain parameter in the EQ effect.

Modulation Bus Destination parameters.



6. **Set the Amount for the second Destination (if desired) by turning the corresponding Amount knob, or by clicking and dragging vertically in the Amount box for the second destination.**
7. **If desired, select a second Destination parameter by click-holding the blue arrow symbol to the right of the corresponding Destination box, and dragging to the desired control on the panel.**
8. **If desired, click the Scale box and select a Scale parameter.**
 The available Scale parameters are the same as the Source parameters, see "Modulation Bus Source parameters.".
9. **Turn the Scale Amount knob, or click the Amount box to the left of the Scale box and move the mouse pointer up or down to set a Scale Amount value.**
 Both positive and negative Scale Amount values can be set (+/- 100%). If you, for example, are using the Mod Wheel as Scale parameter and don't want any modulation when the Mod Wheel is set to zero, set the Scale Amount parameter to 100%. Then, there will be no effect when the Mod wheel is set to zero, and full modulation when the Mod Wheel is all the way up.

- **How much modulation will be applied when the Scale parameter is set to maximum is governed by the to Destination Amount parameter(s).**
- **How much the Scale parameter controls the modulation is set with the Scale Amount parameter.**

→ **To clear an assigned Source, Destination or Scale parameter, hold down [Ctrl](Win) or [Cmd](Mac) and click the Source/Destination/Scale box. Alternatively, click the Source/Destination/Scale box and select "Off" from the list.**

→ **To reset an Amount value to 0, hold down [Ctrl](Win) or [Cmd](Mac) and click the desired Amount box or knob.**

→ **To clear an entire modulation assignment (a whole row), click the circular X button to the right of the corresponding Scale box.**



Connections



! **Remember that CV connections are NOT stored in the Europa patches! If you want to store CV connections between devices, put them in a Combinator device and save the Combi patch.**

Sequencer Control inputs

The Sequencer Control CV and Gate inputs allow you to play Europa from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity. There are also inputs for modulating the Pitch Bend and Mod Wheel parameters.

CV Modulation inputs and outputs

These assignable control voltage (CV) inputs and outputs can be used for modulation of and from assigned Source and Destination parameters in the Modulation Bus section, see "The Modulation Bus section".

Audio Output

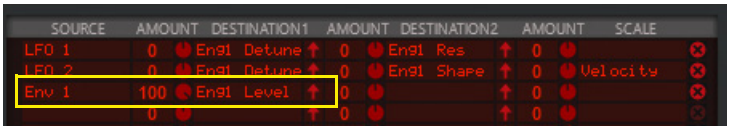
These are the main audio outputs. When you create a new Europa device, these outputs are auto-routed to the first available outputs in the I/O device.

Tips and Tricks

Creating an individual “pre amp envelope” for a Sound Engine

There might be situations where you want to control the amplitude envelopes separately for each Sound Engine. Let’s say you have a plucked sound with a fairly long release in Sound Engine 2 and then you want to slowly fade in a pad sound from Sound Engine 1. Since the built-in Amp Envelope controls all three Sound Engines together, you could use the following “workaround”:

1. **Assign “Envelope 1” as Source and “Mixer: Eng1 Level” as Destination in the Modulation Bus.**
 Turn the Amount knob to 100 in the Modulation Bus.



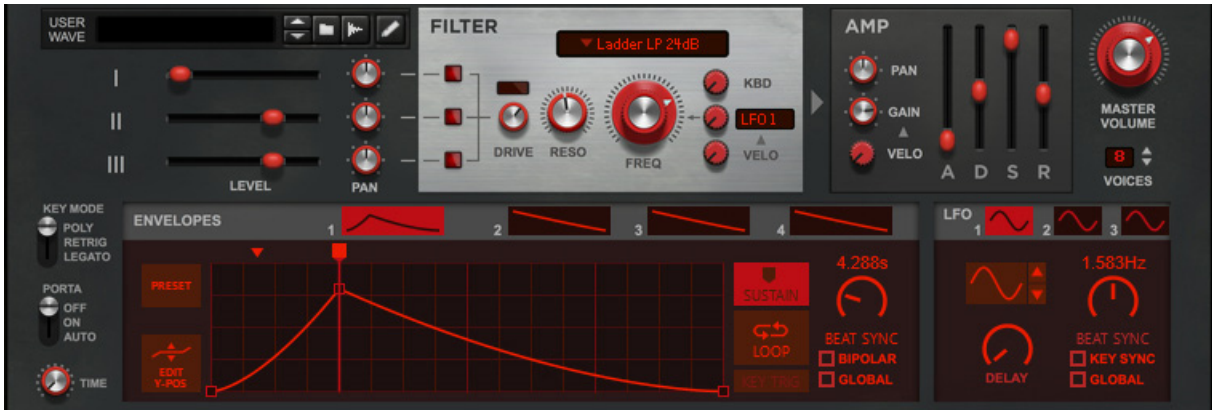
2. **Create your “fading” envelope curve in the Envelope 1 display.**
 Click the Sustain button in the Envelopes display if you want to have a sustain stage in your envelope.



3. **Set the Sound Engine 1 Mixer Level slider to zero.**



4. **As you play the keyboard, Envelope 1 will now fade in the signal from Sound Engine 1, while the signal from Sound Engine 2 is only controlled by the built-in Amp Envelope.**

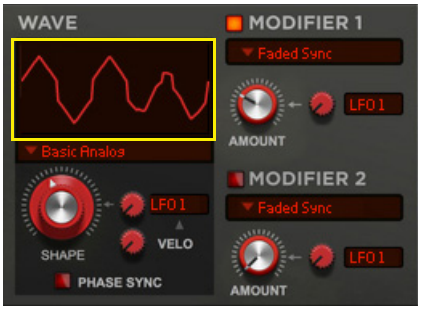


! **Note that the built-in Amp Envelope’s settings will also affect the “fading pad” sound from Sound Engine 1, since all Sound Engine signals eventually pass through the Amp Envelope.**

Recording display movements in the sequencer

If you are in "experimentation mode" and want to try out some wild waveform and/or Spectral Filter tweaking, you can record automation of your interactive display movements in the main sequencer:

1. **Record some notes on the sequencer track in the main sequencer and then hit stop.**
2. **Hit Record again in the main sequencer and click and drag in the Waveform display during recording:**



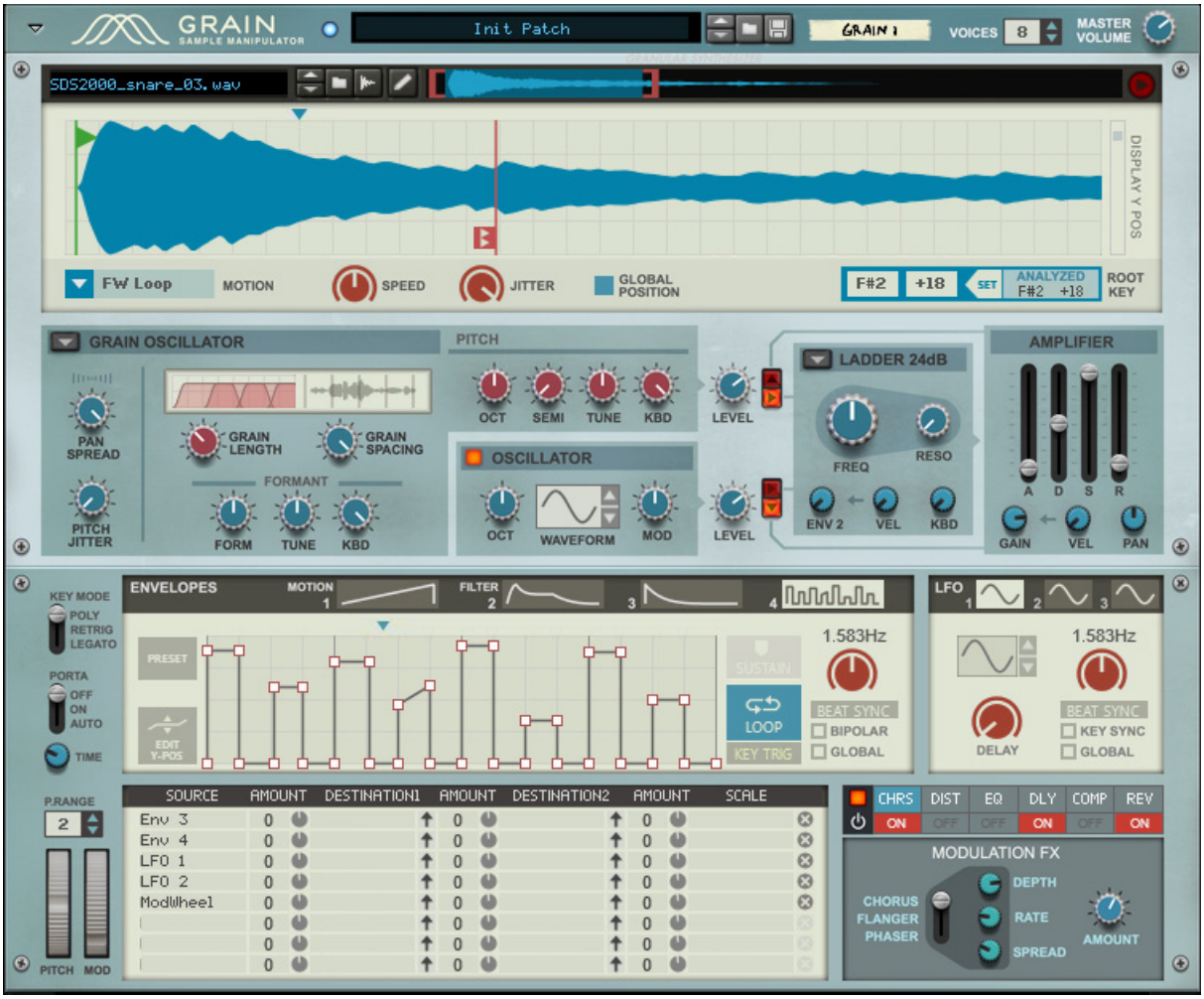
3. **Hit Stop in the sequencer when you are done recording.**
 Any vertical movements have now been recorded as Shape parameter automation and any horizontal movements have been recorded as Modifier 1 Amount parameter automation.
4. **If you like, hit Record again in the Reason sequencer and click and drag in the Spectral Filter display during recording:**



5. **Hit Stop in the main sequencer when you are done recording.**
 Any vertical movements have now been recorded as Spectral Filter Resonance parameter automation and any horizontal movements have been recorded as Spectral Filter Frequency parameter automation.

Chapter 14
Grain Sample Manipulator

Introduction



The Grain Sample Manipulator is a very advanced sampler and granular synthesizer, which offers sonic possibilities far beyond the ordinary. Despite its vast sonic capabilities, Grain has a straight-forward user interface, designed for experimentation.

Grain uses samples as base for sound generation. You could load a sample from your computer and then select various types of sample playback modes and algorithms to manipulate and process the audio. You could also use Grain as a traditional sample player and just play back samples in a regular fashion.

A number of filter and modifier algorithms make it possible to modulate and control the audio further. The extensive Envelopes section and Modulation Bus section allow for detailed and flexible modulation and control. Grain also features a flexible and great-sounding multi-effect to spice up your sounds even more.

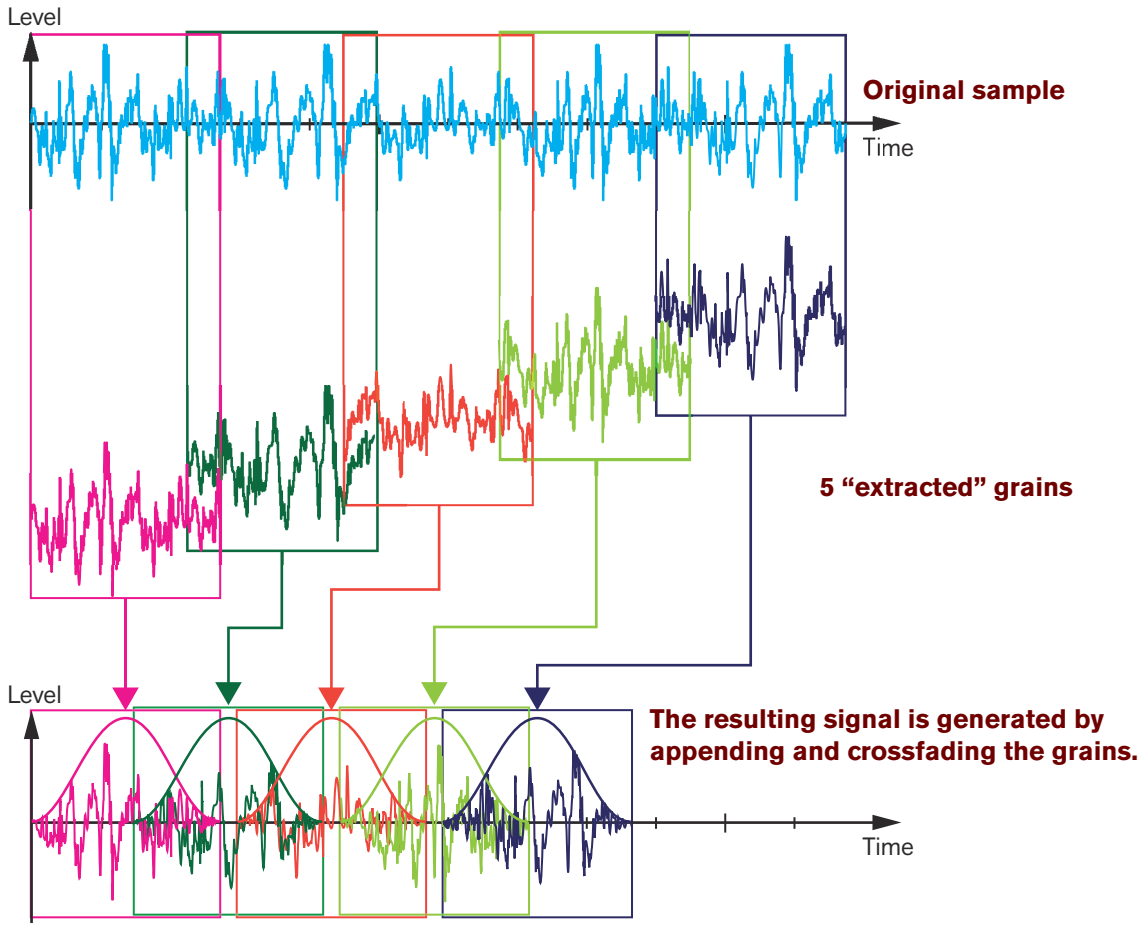
Don't forget to check out the Grain videos here!

! Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.

A few words about granular synthesis

Grain utilizes “granular synthesis” to generate sounds. This synthesis method results in playback of a series of snippets of audio data - grains - “extracted” from an audio sample. The grains could be of a selectable length and spacing, and could be from anywhere in the original sample. The grains could also be played back in a number different ways - with or without crossfades between the grains.

The picture below shows the basic principle of granular synthesis:



An example of a signal generated from 5 grains of a sample.

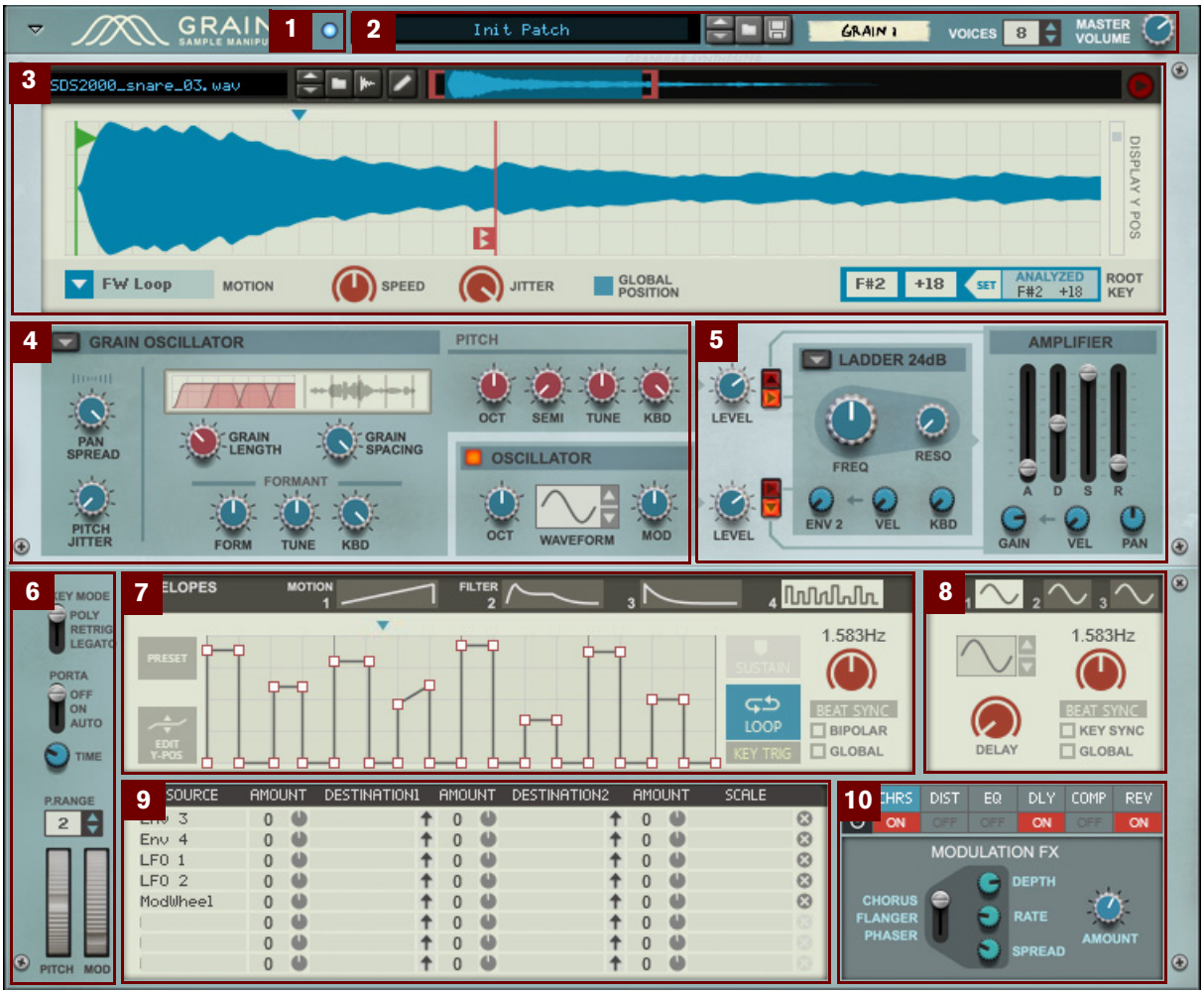
Here is what happens in the example above:

- **The original sample at the top is used as base for the granular synthesis.**
- **5 grains (of the same lengths and the same distances between them) are “extracted” from the original sample.** The distance between the grains is determined by the current sample playback speed. The grains could contain common audio data in some parts (like in the beginnings and ends in the example above).
- **The 5 grains are then placed after one another, partly overlapping each other.** The distance between the grains is determined by the playback rate.
- **When the grains are played back, big parts of the grains are played back together (since they are overlapping).** In the example above, there are also crossfades between the grains to make the overlaps smoother.

Note that the picture above only describes one basic example of granular synthesis - the “Long Grains” playback algorithm in Grain. Grain uses a number of different granular synthesis and spectral synthesis techniques, with different functionality and characteristics.

Panel overview

The Grain front panel contains the following sections:



The Grain front panel sections.

- **1. MIDI Note On LED.**
- **2. Patch Selector (for browsing, loading and saving patches), Polyphony and Master Volume controls.**
- **3. Sample section (for sample loading and sample playback functions).**
- **4. Playback Algorithm and Oscillator section.**
- **5. Filter and Amplifier sections.**
- **6. Global performance and "play" controls.**
- **7. Envelopes section.**
- **8. LFO section.**
- **9. Modulation Bus section.**
- **10. Effects section.**

Playing and using Grain

Loading and saving patches



Loading and saving patches is done in the same way as with any other internal Reason device, see "Loading patches" and "Saving patches" for details.

! **Like with the other sampler devices in Reason, the patch does not include the actual sample - only a reference to it. Therefore, the sample has to be stored separately, or already be on disk or in a ReFill on your computer).**

Global performance and "play" controls



Key Mode

Here you choose how Grain should respond to MIDI Note data:

- **Poly**
 Select this if you want to play Grain polyphonically. The maximum number of voices is 12. The number of voices is set in the Voices control at the top right of the Grain panel, see "Voices".
- **Retrig**
 Select this if you want to play Grain in monophonic mode and always retrigger the envelopes as soon as you play a new note.
- **Legato**
 The Mono Legato mode is also monophonic. However, if you play a new note without having released the previous one, the envelopes and sample playback position won't start over.

► **Also see the description of the "Global Position" parameter. This describes how to play through a sample in a "non-legato" fashion - or polyphonically - in a "sample playback legato" fashion, where new notes will continue at the current sample playback position (and not restart playback).**

Porta

Portamento makes note pitches glide from previous notes to new ones, at the time set with the Time knob. Portamento can be used in all Key modes (see above).

- **When On in Poly Key Mode (see above), the pitches will glide from any of the available voices.**
 The results will be unpredictable since there is no way of controlling from which note(s) the glide(s) will commence. The effect is very nice, though.
- **When On in Retrig or Legato Key Mode (see above), the pitch will glide between consecutive notes.**
- **In Auto mode, the pitch will glide between consecutive monophonic notes only when you play legato. If you have selected Poly Key Mode (see above), Auto will have no effect at all.**
 If you release the previous key before hitting the new key, there will be no portamento effect.

Range

→ **Set the desired Pitch Bend range for the "Pitch" wheel with the up/down buttons, or by click-holding on the display and dragging up/down.**
Range: +/-24 semitones (+/-2 octaves) in steps of +/-1 semitone.

Pitch

The Pitch bend wheel can be used for bending note pitches up and down. Grain also responds to Pitch Bend MIDI data from a connected MIDI master keyboard. You set the desired Pitch bend Range with the "Range" control above the Pitch bend wheel.

! **Note that with some playback algorithms, such as Spectral Grains, the audible pitch depends on the formant rather than the pitch settings (see "Spectral Grains"). For pitch bend to have an effect here, you need to add a Pitch Wheel -> Formant routing in the Modulation Bus, see "The Modulation Bus section".**

Mod

The Mod wheel can be used for controlling almost any parameter in Grain. Use the Mod wheel as a Source parameter in the Modulation Bus section and then route to the desired Destination parameter(s), see "The Modulation Bus section".

Global output controls



Voices

Here you set the polyphony of your patch, from 1 to 12 voices.

► **If you want monophonic playback you can use the "Retrig" and "Legato" modes instead of lowering the Voices parameter to 1.**

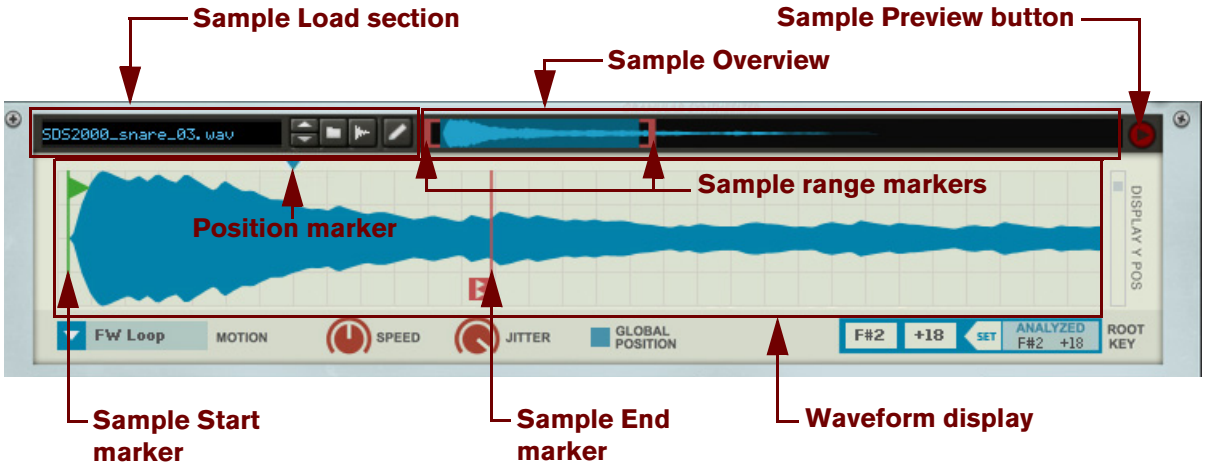
Master Volume

This is the main stereo output volume control.

Panel reference

The Sample section

Here is where you load/sample and configure the audio that should serve as the base for the granular synthesis.



Loading

- **Load a sample using drag & and drop, or by clicking the Browse sample button, or by using the Up/Down buttons to scroll and load a sample from the currently selected folder.**

! **It's possible to load stereo samples. However, the waveform will always be displayed as a mono signal, regardless if it's mono or stereo.**

Setting the sample range

First you could decide how much of the original sample you want to use - and where in the sample you want to work:

- **Zoom and/or scroll in the Sample Overview to define the Sample range you want to work in.**
 To scroll, click and drag between the orange sample range markers. To zoom, click and drag any of the sample range markers sideways. The set Sample range is automatically updated and displayed in the waveform display.
- **To work in the entire Sample range, drag the left Sample range marker all the way to the left, and the right Sample range marker all the way to the right, in the Sample Overview.**

– **It's possible to automate the sample range settings, see "Automating sample playback parameters from the sequencer".**

Setting the sample start and end

→ **Drag the green Sample Start marker to where you want the sample to begin playing back.**

→ **Drag the red Sample End marker to where you want the sample playback to end.**
The green triangular "flag" on the Sample Start marker shows the current playback direction. If the Sample Start marker should be to the right of the Sample End marker, the sample will play back in the opposite direction.

! **Note that the Freeze playback motion mode don't have any Sample End Marker, see "Freeze" below.**

⌐ **It's possible to automate the sample start and end settings, see "Automating sample playback parameters from the sequencer".**

Motion

Motion controls the way the Position marker ("playhead") is played back in the original sample. The Motion modes work in conjunction with the Sample Start/End markers in the waveform.

→ **Click the Motion selector to choose one of the following playback motion modes:**

- **Freeze**
 In Freeze mode, the sample is played back at (and around) the Sample Start marker position. There is no Sample End marker in this mode. Note that if you have selected the Tape algorithm (see "Tape"), there will be no sound.
- **One Shot**
 In One Shot mode, the sample is played back (from the Sample Start marker to the Sample End marker) in its entirety each time you press a key.
- **FW Loop**
 In FW Loop mode, the sample is looped forward (from the Sample End marker to the Sample Start marker) for as long as you hold down a key.
- **FW-BW Loop**
 In FW-BW Loop mode, the sample is looped back and forth between the Sample End marker and the Sample Start marker for as long as you hold down a key.
- **End Freeze**
 In End Freeze mode, the sample is played back once from the Sample Start marker to the Sample End marker and then played back at (and around) the Sample End marker position. Note that if you have selected the Tape algorithm (see "Tape"), there will be no sound after you reached the Sample End marker.
- **Envelope 1**
 In Envelope 1 mode, the sample is played back between the Sample Start marker and the Sample End marker according to the Envelope 1 curve (see "The Envelopes section"). The Sample Start position is represented by the minimum Y value and the Sample End position is represented by the maximum Y value in the Envelope display.

 The Envelope 1 mode is also the mode to use if you want to play back and loop the sample in sync with the Reason sequencer. Use a straight ramp (up) in Envelope 1, activate Beat Sync and set the sync to a suitable bar length, see "Looping the envelope".

Speed

The Speed control determines how fast the play position moves in the waveform.

→ **Set the sample playback speed with the Speed knob.**
Depending on which Motion mode and Playback Algorithm is currently selected, the sonic result may vary heavily. If you have selected the Tape algorithm (see "Tape"), the Speed knob also affects the pitch. Note that the Speed can be set all the way down to 0%, i.e. "stop". Great for Tape Stop effects, for example.

! **Note that the Speed control doesn't have any effect when you use the Envelope 1 motion mode, see "Envelope 1" above.**

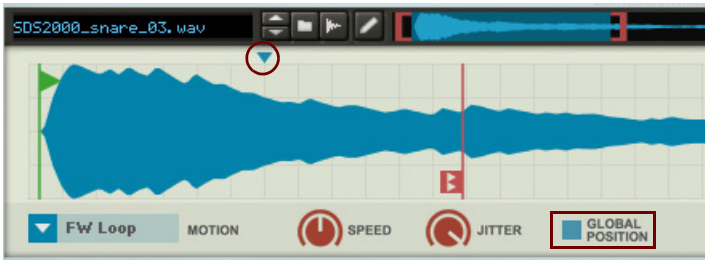
Jitter

The Jitter function modulates the sample playback position minutely and randomly. The Jitter function can be great for generating “chorus”-like effects and to make a sound more “alive”, depending on the other settings in the sound.

→ **Set the playback position deviation with the Jitter knob.**
At 0%, the timing and playback position is completely accurate and at 100% it is completely random.

Global Position

→ **Click the Global Position button to start playback of new voices from the global position, i.e. from where the blue Position marker is currently positioned in the waveform display:**



This function is great for rhythmic and vocoder-like sounds etc.

If not active, new voices will always start playing back from the Sample Start marker.

Root Key

A sample is automatically analyzed for its original pitch at the Sample Start position. The analyzed pitch is displayed in the Analyzed display in the Root Key section. If you move the Sample Start marker, the sample is automatically re-analyzed.

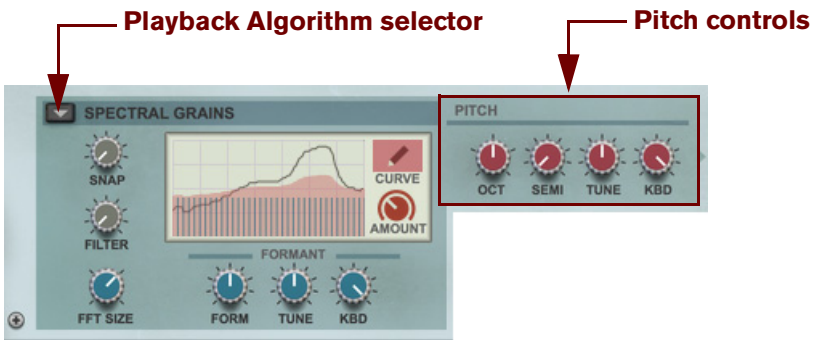
→ **Click the “SET” button to use the analyzed Root Key.**
This will automatically place the analyzed Root Key on the correct note in the keyboard range.

→ **Alternatively, define the Root Key manually by dragging up/down in the “Semitone” and “Cents” boxes.**

The Playback Algorithms section

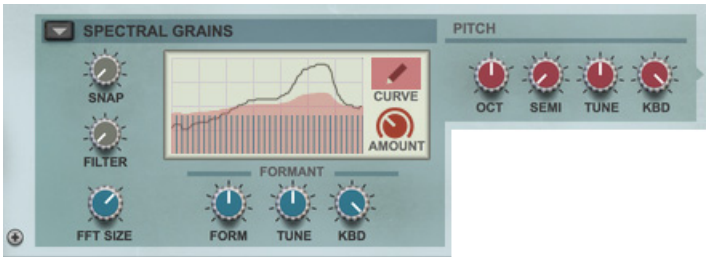
Here is where you select which Playback Algorithm to use for manipulating the sample. Each of the Playback Algorithms produce very different sonic results and have different unique parameters.

! **Note that the common Pitch controls to the upper right could affect the sound differently, depending on the selected algorithm.**



→ **Click the Playback Algorithm selector and choose one of the following four algorithms:**

Spectral Grains

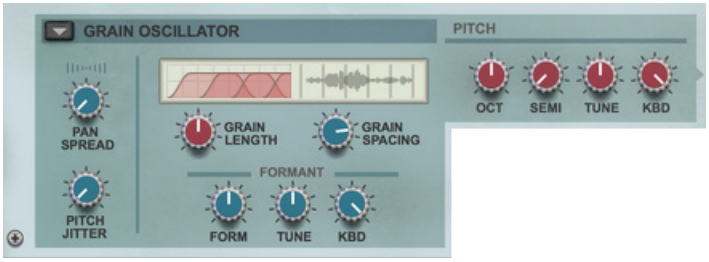


The Spectral Grains playback algorithm uses FFT analysis to analyze the frequency content (partials) of the original sample. You can then stretch the generated signal by pitch-shifting the partials, and also filter out inharmonic partials. This way you could continuously transform inharmonic signals into harmonic signals, for example. You can also draw your own formant curves in the spectrum display to give the sounds different pitches/characters.

- **Snap**
 This pitch-shifts inharmonic partials towards the closest harmonic partials. At 0% the sound is almost unaffected and at 100% the sound contains only harmonic partials.
- **Filter**
 Instead of pitch-shifting inharmonic partials towards harmonic ones, as the Snap control above does, the Filter control filters out the inharmonic partials and keeps the harmonic ones. Since the filter slopes are not brickwall shaped some of the inharmonic partials (if any) will remain audible even at 100%.
- **FFT Size**
 This sets the accuracy (and speed) of the frequency analysis. "0" is the fastest detection, but this also leaves out detection of low frequencies. "3" is the most accurate detection. However, it's also slower since it also detects low-frequency material (which takes longer to detect).
- **Curve**
 With the Curve tool you can draw your own formant curves in the frequency spectrum. Drawing above the pink area means the partials are amplified, and drawing below the pink area means the partials are attenuated.
- **Amount**
 Set the gain/attenuation amount of the drawn formant curve (see "Curve" above). At 0% the curve is completely flat.
- **Formant**
 Sets the initial pitch of the sample, together with the Root Key setting (see "Root Key"). If Snap and Filter are both set to 0%, the Root Key and Formant controls the pitch of the signal. This also means that the Pitch parameters (see "Pitch controls") and Pitch wheel (see "Pitch") have no effect. To have the Formant track the keyboard in a musical way, make sure the Formant Kbd parameter (see below) is set to 100%.

 When you raise the Snap or Filter parameters towards 100% the sound gradually adapts to the Pitch settings instead, and the Root Key and Formant parameters now affect the tone color instead.
- **Formant Tune**
 Here you can fine-tune the formant curve to adjust the pitch to the Oscillator pitch (see "The Oscillator section").
- **Formant Kbd**
 Here you set how much you want the formant to track the keyboard. 0% means no keyboard tracking and 100% means full 1:1 keyboard tracking. If the Snap and Filter parameters (see above) are both set to 0%, make sure the Formant Kbd is set to 100% to make the audible pitch track the keyboard one semitone per note.

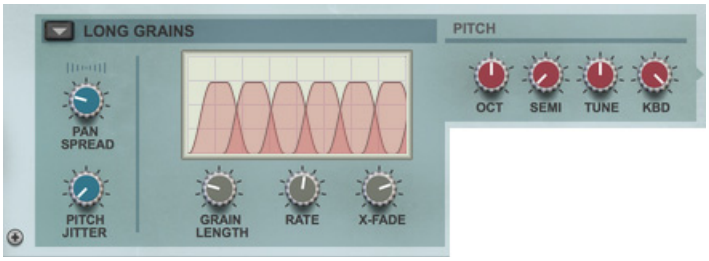
Grain Oscillator



The Grain Oscillator plays back a mix of two very short grains of the original sample. The grain playback rate corresponds to the oscillator pitch. This means the original pitch (Root Key/Formant) of the sample doesn't affect the pitch of the sound, but the timbre.

- **Pan Spread**
 Here you set how much you want the grains to be panned in the stereo panorama. 0% means the signal will be unaffected and 100% means every other grain will be panned hard left and hard right. Great for nice stereo effects and for the impression of an added stereo sub-oscillator, depending on the settings. Note that the pitch of the panned signal becomes 1 octave lower than the original signal due to the fact that every other grain is panned.
- **Pitch Jitter**
 Changes the pitch of every grain. The pitch modulation character is "smooth random".
- **Grain Length**
 Sets the lengths of the grains and also the crossfade amount. At 0% you get the shortest grains and almost no crossfade at all. This means the sound could be a little gritty at this setting. At 100% you get longer grains, that also overlaps each other with a smooth crossfade.
- **Grain Spacing**
 Sets the spacing in the original sample between the two played back grains. High Spacing values render more even sound character throughout the played notes - almost like a wavetable synth - since a lot of audio data in the original sample is skipped. Less spacing normally creates more varying sound character between each played note.
- **Formant**
 Sets the formant's initial frequency. Turn this knob to change the tone color of the sound. At high Grain Spacing values (see above) the effect of changing the Formant could be similar to the classic "oscillator sync" sound. To have the Formant fully track the keyboard, make sure the Formant KBD parameter (see below) is set to 100%.
- **Formant Tune**
 Here you can fine-tune the formant curve.
- **Formant Kbd**
 Here you set how much you want the formant to track the keyboard. 0% means no keyboard tracking and 100% means full 1:1 keyboard tracking.

Long Grains

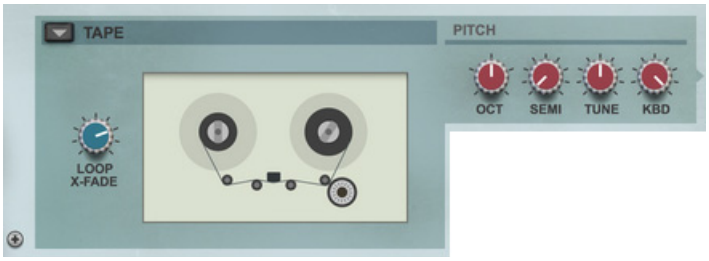


The Long Grains playback algorithm plays back fairly long grains of the original sample. This means that it's the original pitch of the sound (Root Key) that affects the pitch, along with the Pitch settings (see "Pitch controls").

The display shows the effects of the Grain Length, Rate and X-Fade settings.

- **Pan Spread**
 Here you set how much you want every other grain to be panned in the stereo panorama. 0% means the signal will be unaffected and 100% means every other grain will be panned hard left and hard right. Great for nice stereo effects!
- **Pitch Jitter**
 Changes the pitch of every grain. The pitch modulation character is "smooth random".
- **Grain Length**
 Sets the lengths of the grains. At 0% you get the shortest grains and towards 100% you get longer grains.
- **Rate**
 This controls the playback rate of the grains.
- **X-Fade**
 Here you set the crossfade between the grains. At 0% there is minimal crossfade, which will give the signal a gritty or "popping" character at the playback start and end of each grain.

Tape



The Tape playback algorithm plays back the sample the old-fashioned "tape-style" way, where playback speed and pitch are linked. If playback speed is zero (in Freeze and End Freeze Motion modes for example), no sound will be heard - but you can drag, modulate or automate the playback position for scrubbing and tape stop effects.

- **Loop X-Fade**
 Sets the crossfade amount if you have selected FW Loop or FW-BW Loop as Motion type (see "Motion")

! **Note that the Loop X-Fade control has no effect if you have selected "Envelope 1" as Motion type.**

! **If you have selected "Envelope 1" as Motion type (see "Envelope 1"), the Speed (see "Speed") and Pitch settings (see "Pitch controls") have no effect. The sample will play back at the same pitch regardless of which note you play.**

Pitch controls



- **OCT**
 Sets the pitch in octave steps.
 Range: 5 octaves.
- **SEMI**
 Sets the pitch in semitone steps.
 Range: 12 semitones (one octave).
- **TUNE**
 Changes the pitch in steps of 1 cent.
 Range: +/- 50 cents (down or up half a semitone).
- **KBD**
 Sets how much the pitch should track incoming MIDI Notes.
 Range: 0% (no tracking (constant pitch)) to 100% (1 semitone per key).

! **In the Spectral Grains playback algorithm (see "Spectral Grains"), the Pitch controls have no effect if Snap and Filter are set to 0%. To get full effect of the Pitch controls, set Snap or Filter to 100%.**

The Oscillator section



The Oscillator can be used in addition to the sample playback. The oscillator features a number of selectable waveforms and a modulation control, which affect the signals differently depending on selected waveform. The oscillator pitch always tracks the keyboard to 100%. This makes it perfect as a pitch reference for the sample signal.

On/Off

→ **Click the On/Off LED button to switch on/off the oscillator.**

Oct

→ **Turn the OCT knob to change the pitch in octave steps.**
Range: 5 octaves.

Waveform and Mod

→ **Select the desired waveform by dragging up/down in the waveform display, or by clicking the up/down buttons.**
The waveforms are:

- **Sine**
 A pure sinewave at Mod=0%, gradually transformed towards a sawtooth type signal at Mod=100%.
- **Triangle**
 A triangle wave at Mod=0%, gradually transformed towards a sawtooth type signal at Mod=100%.
- **Sawtooth**
 A lowpass-filtered sawtooth wave at Mod=0%, gradually transformed to a pure sawtooth signal at Mod=100%.
- **Square/Pulse**
 A 50% duty cycle square wave at Mod=0%, gradually pulsewidth-modulated to a 0% duty cycle pulse wave (silence) at Mod=100%.

← **Modulate the Mod parameter from an LFO (using the Modulation Bus) to achieve PWM.**

- **Noise**
 A level-compensated lowpass-filtered noise at Mod=0%, gradually transformed to white noise at Mod=100%.
- **Band Noise**
 A noise-modulated sinewave. The Mod knob controls noise bandwidth. At Mod=100%, the oscillator produces pure noise. Turning the knob counter-clockwise towards 0% gradually narrows the noise bandwidth until a slightly modulated sinewave is produced.

The Filter section



The signals from the Playback Algorithms section and the Oscillator section can be individually mixed and routed through the Filter section. The Filter section features four different filter types.

Routing buttons

→ **Click the red buttons with a triangle pointing to the right, to route the desired signal to the Filter section.**
To bypass the signals from the Filter section, click the buttons with the triangle pointing upwards or downwards.

Filter type

→ **Click the Filter type selector to choose any of the following filter types:**

- **HP 12dB**
 A highpass filter with a 12dB/octave slope.
- **BP 12dB**
 A bandpass filter with 12dB/octave slopes.
- **LP 12dB**
 A lowpass filter with a 12dB/octave slope.
- **LP Ladder 24dB**
 A ladder-type lowpass filter with a 24dB/octave slope.

Freq

→ **Set the cutoff frequency (for the HP and LP filter types) or the center frequency (for the BP filter type).**

Reso

→ **Set the resonance amount.**

Env 2

→ **Set the cutoff/center frequency modulation amount from the Envelope 2.**
Since this is a “hardwired” connection from Envelope 2 you don't need to use the Modulation Bus for envelope modulating the cutoff/center frequency.

Vel

→ **If you want the Envelope 2 amount to be controlled from keyboard velocity, turn up the Vel knob.**

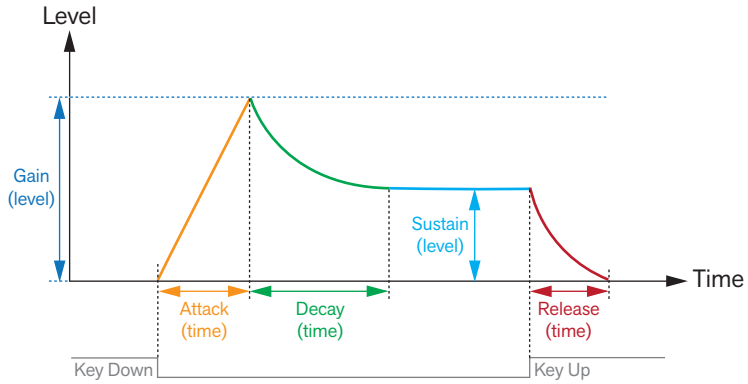
Kbd

→ **Set how much you want the filter cutoff/center frequency to track the keyboard.**
At 0%, the filter frequency is static regardless where on the keyboard you play. At 100% the filter tracks the keyboard 1:1, i.e. one semitone per note.

The Amplifier section



The Amplifier section contains a standard ADSR envelope which controls the amplitude of the signals from the Playback Algorithms and Oscillator sections equally. The picture below shows the various stages of the ADSR envelope:



The ADSR envelope stages.

A(ttack)

When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the value set with the Gain knob (see below). How long this should take, depends on the Attack setting. If the Attack is set to "0", the Gain level is reached instantly. If the Attack value is raised, it will take longer time before the Gain level is reached.

D(ecay)

After the Gain level has been reached, the level starts to drop. How long this should take is governed by the Decay parameter.

If you want to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0", the Decay parameter should be set to a medium value and the Sustain level should be set to "0", so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you raise the Sustain parameter.

S(ustain)

The Sustain parameter determines the level the envelope should rest at, after the Decay stage. If you set Sustain to full level, the Decay setting is of no importance since the volume of the sound is never lowered.

If you want to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

But often a combination of Decay and Sustain is used to generate envelopes that rise up to the Gain level, then gradually decreases to finally land to rest on a level somewhere in-between zero and the Gain level. Note that Sustain represents a level, whereas the other envelope parameters represent times.

R(elease)

The Release parameter works just like the Decay parameter, except it determines the time it takes for the volume to drop back to zero after you release the key.

Gain

→ **Set the desired maximum level for the Amplifier with the Gain knob.**
This is the maximum level the envelope will reach after the Attack stage is completed (see above).

← **If you want to create a tremolo effect, assign "Gain" as Destination and an LFO as Source in the Modulation Bus section, see "The Modulation Bus section".**

Vel

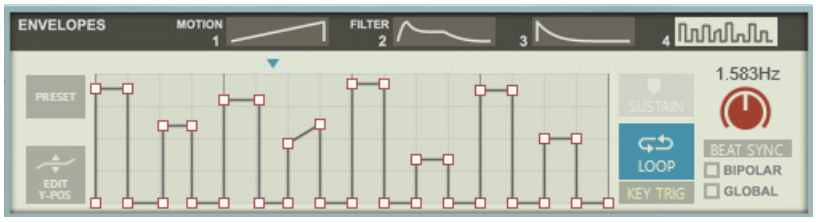
→ **If you want the Gain level to be controlled from keyboard velocity, turn up the Vel knob.**

Pan

→ **Set the panning of the output signal from the Amplifier in the stereo panorama.**

← **Since Pan works individually per voice, you can assign e.g. Keyboard Velocity or an Envelope in the Modulation Bus to control the Pan effect, see "The Modulation Bus section".**

The Envelopes section



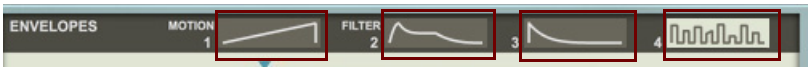
The Envelopes section features four separate polyphonic (one per voice) general purpose envelope generators, that can be assigned to control selectable parameter(s) in the Modulation Bus section. The first two envelopes (Envelope 1 and Envelope 2) are also hardwired to the Motion and Filter Frequency destinations respectively.

The Envelopes are extremely flexible, and you can draw your own custom modulation shapes by clicking and drawing in the display area. There are also a number of preset shapes that you can use as starting points (or use as is). If you use Loop mode, you could turn the envelope into a kind of LFO.

See "The Modulation Bus section" for details on how to assign the Envelopes to the desired destination(s).

Envelope 1, 2, 3 and 4

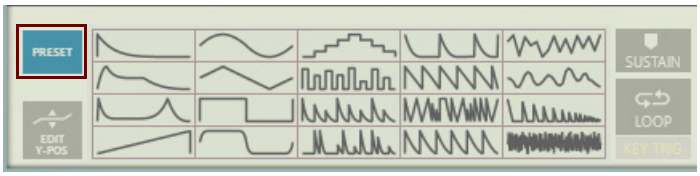
→ **Click one of the Envelope 1, 2, 3 or 4 buttons to select which envelope to edit:**



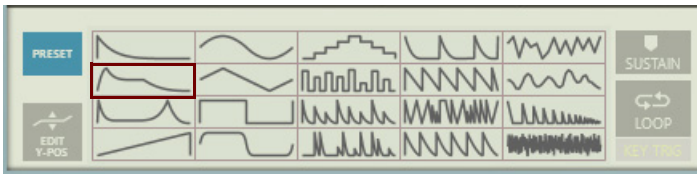
Envelope 1 and Envelope 2 are also hardwired to the Motion and Filter Frequency destinations respectively.

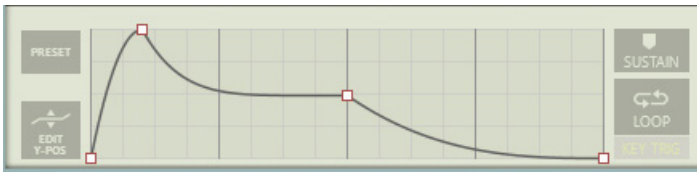
Preset

1. **Click the Preset button to bring up a palette of envelope preset curves:**



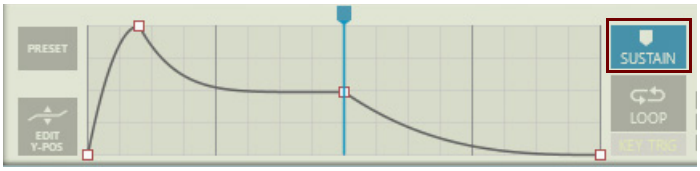
2. **Click the desired envelope preset curve to place it on the display.**
 Let's select a standard ADSR style of envelope curve:





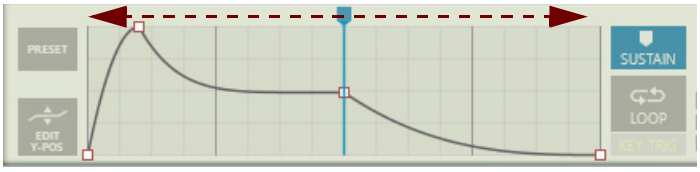
Adding a Sustain stage

→ **Click the Sustain button to add a sustain stage to the envelope:**



The vertical blue marker that appears indicates where the envelope will stay sustained until you release the key.

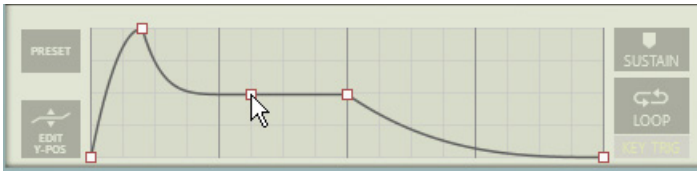
→ **Drag the sustain marker sideways to move the sustain stage to the desired position:**



→ **To remove the sustain stage, click the Sustain button.**

Adding and removing envelope points

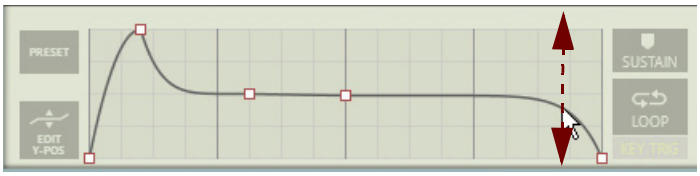
→ **Double click, or hold down [Ctrl](win) or [Cmd](Mac) and click, in the envelope display to add points to the envelope curve:**



→ **To remove a point, double click, or hold down [Ctrl](win) or [Cmd](Mac) and click, on an existing point on the envelope curve.**

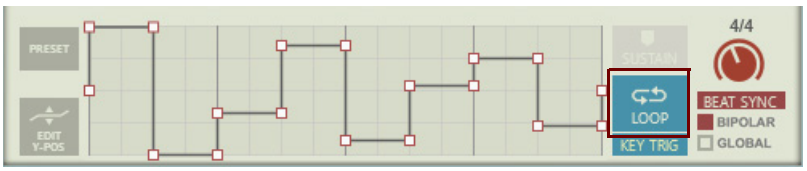
Changing the envelope curve shape

→ **Click a line segment (between two points) and drag up/down to change the curve shape:**



Looping the envelope

→ **Click the Loop button to turn the envelope into a kind of LFO.**
If there was previously a sustain stage in the envelope, this will automatically be disabled when you click the Loop button.

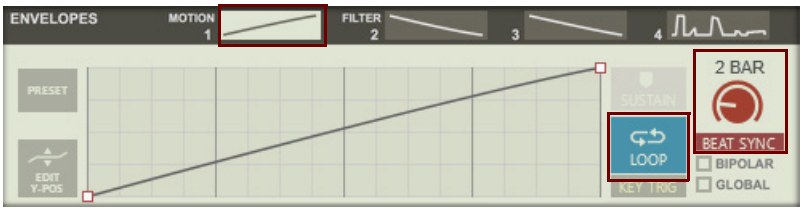


Here we have edited a stepped curve from the Presets. We have also enabled Beat Sync and set the length/rate to 4/4. This means that each step in the curve now represents an 1/8th note.

- **Key Trig means the envelope restarts when you play a note.**
- **You can choose whether the envelope should send out a bipolar value or unipolar one (0-100%).**
- **If Global is on, the envelope will be common for all voices.**

Another useful application for looped envelopes is to sync the sample playback to the Reason sequencer when using the Envelope 1 Motion mode (see "Envelope 1"):

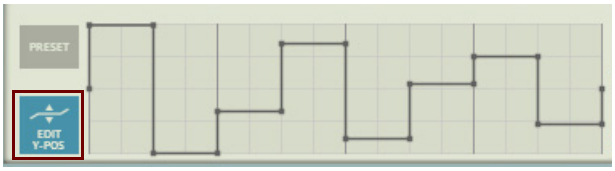
1. **Select Envelope 1 (since it is hardwired to the sample playback Motion parameter).**
2. **Select the "Ramp Up" Preset, enable Loop and set the Beat Sync to the desired value:**



Playing back Reason's sequencer now plays back the sample synced to the sequencer Tempo.

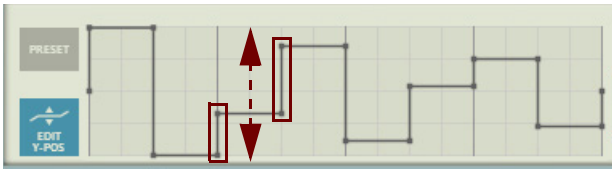
Editing levels only

→ **To restrict the editing to levels only, without affecting the time positions, click the Edit Y-Pos button:**



In this mode you cannot change the time positions of the envelope points, only their levels (height). This is extra useful with a stepped Preset curve, because dragging up or down will change the value of an entire segment, turning the Envelope into a pseudo-sequencer.

! **To be able to adjust the level of a segment, the two points on either side of the segment have to be on the exact same time positions. Otherwise, only the closest point will be changed. Also, any inclining/declining segment will automatically turn horizontal when edited:**

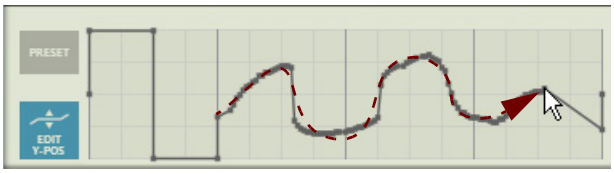


Adjusting the level of a segment.

Creating “free form” envelope curves

As a special feature in the Edit Y-Pos mode, you can also draw “free form” curves:

→ **To continuously add new consecutive points, hold down [Ctrl](win) or [Cmd](Mac) and drag in the envelope display:**



→ **To erase points, hold down [Shift] and [Ctrl](win) or [Cmd](Mac) and drag in the envelope display.**

The LFO section



An LFO (Low Frequency Oscillator) is used for generating cyclic modulation. A typical example is to have an LFO modulate the pitch of a signal to produce vibrato, but there are countless other applications for LFOs.

The LFO section features three separate general purpose LFOs, that can be assigned to control selectable parameter(s) in the Modulation Bus section.

- **Select which of the three LFOs you want to edit by clicking one of the LFO 1, LFO 2 and LFO 3 buttons.**
- **Select an LFO waveform by clicking the spin controls to the right of the waveform display, or by click-holding in the display and dragging up or down.**
 Besides the standard waveforms (sine, triangle, pulse, etc.) there are random, slope and stepped waveforms. The shape of the waveforms are shown in the display.
- **Set the LFO frequency with the Rate knob.**
- **Click the Beat Sync button to sync the LFO to the main sequencer tempo.**
 The Rate parameter now controls the time divisions.
- **Click the Key Sync button to restart the LFO at every new Note On.**
- **Click the Global button to make the LFO common for all voices (monophonic).**
- **Turn the Delay knob to introduce a delay before the LFO modulation kicks in after a note is played.**
 Turn clockwise for longer delay times.

The Effects section



The Effects section features six different effect modules that can be freely reordered by dragging & dropping. Most of the effect parameters are also available as destinations in the Modulation Bus, see "The Modulation Bus section".

At the top of the Effects section are six Effect buttons. Click any of these to bring up the control panel for the corresponding effect. Below the Effect buttons are the On/Off buttons for the individual effects. Click these to activate the effects.

Reordering the effects

- **To define the order of the effects in the serial connection, click and hold on the desired Effect button and drag to the desired position:**



Moving the Chorus effect to another position in the effects chain.

You can reorder the effects at any time.

Phaser/Flanger/Chorus



This is a stereo Phaser/Flanger/Chorus.

→ **Select effect type with the Phaser/Flanger/Chorus switch.**
The selected effect type is displayed on the Effect button.

- **Depth**
Sets the depth of the selected effect. To get a static sound, set Depth to zero.
- **Rate**
Sets the rate/speed of the modulation.
- **Spread**
Sets the stereo width of the effect.
- **Amount**
Sets the Dry/Wet amount of the effect.

← **It's also possible to modulate the start/center frequency of the Phaser/Flanger/Chorus using the "Chorus/Flanger/Phaser > Frequency" destination in the Modulation Bus, see "The Modulation Bus section".**

Distortion



The Distortion effect features six different types of distortion.

→ **Select distortion type with the switch.**
"Dist" produces a dense, rich analog type of distortion.

"Scream" produces a less bright type of distortion.

"Tube" emulates a tube type of distortion.

"Sine" is a sine shaping distortion.

"S/H" gives the effect of sample rate reduction.

"Ring" is a ring modulator effect.

- **Drive**
Sets the overdrive/feedback level of the selected distortion.
- **Tone**
This is a lowpass filter and sets the tone of the selected distortion.
- **Amount**
Sets the Dry/Wet amount of the distortion.

EQ



The EQ effect is a single band parametric equalizer with adjustable Q-value and Gain.

- **Freq**
 Sets the center frequency of the EQ band.
- **Q**
 Sets the bandwidth of the EQ band, from wide to narrow.
- **Gain**
 Sets the gain/attenuation of the EQ band, from -18dB to +18dB.

Delay



This is a stereo delay, routed as a send effect.

- **Sync**
 Activate Sync to sync the delay time to the main sequencer tempo.
- **Time**
 This sets the time between the delay repeats. If Sync is active (see above), the Time parameter now controls the time divisions.
- **Ping Pong**
 Activate Ping Pong to have the delay repeats alternating between left and right in the stereo panorama. The effect is also dependent on the Pan parameter (see below).
- **Pan**
 Sets the panning of the delay repeats in the stereo panorama. If Ping Pong is active (see above) the Pan knob controls the panning of the initial delay repeat as well as the total stereo spread of the remaining repeats.
- **FB**
 The FB (feedback) parameter determines the number of delay repeats.
- **Amount**
 Use this parameter to adjust the send level to the Delay effect.

► **If you play a note, have a long delay feedback and turn down Amount, the echoes will continue. This allows for automated "triggered delay" fx.**

Compressor



This is a stereo compressor.

- **Attack**
 This governs how quickly the compressor will apply its effect when signals rise above the set threshold. If you raise this value, the response will be slower, allowing more of the signal to pass through the compressor unaffected. Typically, this is used for preserving the attacks of the sounds.
- **Release**
 When the signal level drops below the set threshold, this determines how long it takes before the compressor lets the sound through unaffected. Set this to short values for intense, "pumping" compressor effects, or to longer values for a smoother change of the dynamics.
- **Thres**
 This is the threshold level above which the compression sets in. Signals with levels above the threshold will be affected, signals below it will not. In practice, this means that the lower the Threshold setting, the more the compression effect.
- **Ratio**
 This specifies the amount of gain reduction applied to the signals above the set threshold.

Reverb



This is a stereo reverb, routed as a send effect.

- **Decay**
 This governs the length of the reverb effect.
- **Size**
 Sets the emulated room size, from small room to large hall. Middle position is the default room size.
 Lowering this parameter results in a closer and gradually more "canned" sound. Raising the parameter results in a more spacey sound, with longer pre-delay.
- **Damp**
 Raising the Damp value cuts off the high frequencies of the reverb, thereby creating a smoother, warmer effect.
- **Amount**
 Use this parameter to adjust the send level to the Reverb effect.

 If you play a note, have a long delay Decay time and turn down Amount, the reverberation will continue.

The Modulation Bus section



The Modulation Bus section is used for routing a modulation Source to one or two modulation Destinations each. This creates a very flexible routing system that complements the pre-wired routing in Grain.

The Modulation Bus section in Grain is derived from the one in the Reason Thor Polysonic Synthesizer device, so if you are familiar with Thor, you will quickly find your way around in Grain's modulation bus.

There are eight "Source –> Destination 1 –> Destination 2 –> Scale" busses, of which the first four have pre-assigned sources. However, these four pre-assigned sources can be easily changed if you like.

A Source parameter can modulate two different Destination parameters per bus (with variable Amount settings). Each bus also has a Scale parameter that affects the relative modulation Amount for both Destinations.

- **Note that it is possible to assign the same source parameter as Source in several busses. This allows you to control more than two Destination parameters from the same Source.**

1. Select the desired Source parameter by clicking in the corresponding Source box and selecting from the list.

The following parameters can be used as modulation Sources:

Parameter	Description
Velocity	This applies modulation according to the Keyboard Velocity values (how hard or soft you strike the MIDI keyboard keys).
LFOs (LFO 1, LFO 2 and LFO 3)	This allows you to modulate parameters from LFO 1, LFO 2 and LFO 3 respectively.
Envelopes (Amp Envelope, Envelope 1, Envelope 2, Envelope 3, Envelope 4, Envelope 3 * Envelope 4, Envelope 3 * LFO 3)	This allows you to modulate parameters from any of the Envelopes. As a special feature you can also modulate parameters from the multiplied signal of Envelope 3 and Envelope 4, as well as from the multiplied signal of Envelope 3 and LFO 3.
Mod Wheel	This allows you to modulate parameters from the Mod Wheel.
MW Latched	This allows you to modulate parameters based on the current Mod Wheel value at a given Note On.
Pitch Wheel	This allows you to modulate parameters from the Pitch Bend control.
Breath	This allows you to modulate parameters from the Breath performance controller
Expression	This allows you to modulate parameters from the Expression performance controller
Aftertouch	This allows you to modulate parameters from Keyboard Aftertouch (channel aftertouch)
Sustain	This allows you to modulate parameters from a connected sustain pedal. Note that continuous sustain data (0-127) is supported - not just on/off.
Key	This is keyboard tracking. If a positive Amount value is used and the destination is filter frequency, the filter frequency will track the keyboard, i.e. increase with higher notes.
Random	This sends out a random value each time a new note is played.
Key In Octave	This allows you to modulate parameters based on 12 separate note values (within each octave).
Noise	This allows you to modulate parameters from white noise.
Polyphony	This allows you to modulate parameters based on the number of playing voices at a given time.
Last Velocity	This applies modulation according to the latest Keyboard Velocity value (how hard or soft you hit the latest MIDI keyboard key).
Sample Pitch Curve	As soon as you load a sample in Grain the pitches throughout the entire sample is automatically analyzed and saved as a "Pitch Curve". This allows you to modulate parameters based on the analyzed pitch value at the Position marker's current position in the original sample.

Parameter	Description
Display Y Position	This allows you to modulate parameters based on the mouse pointer's Y position in the sample window. See "Automating sample playback parameters from the sequencer" for an example.
Display Mouse Gate	This allows you to modulate parameters based on the clicking/holding the mouse in the sample window. See "Automating sample playback parameters from the sequencer" for an example.
CV Input 1/2/3/4	This takes the current value on the CV 1/CV 2/CV 3/CV 4 inputs on the rear panel and sends to the desired destination.
CV Input 1/2/3/4 Latched	This allows you to modulate parameters based on the current CV 1/CV 2/CV 3/CV 4 value at a given Note On.

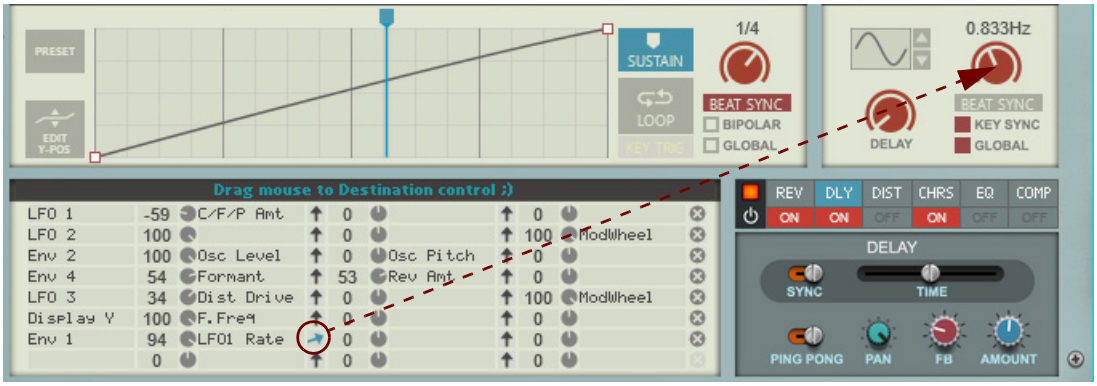
Modulation Bus Source parameters.

2. **Set the Amount for the first Destination by turning the corresponding Amount knob, or by clicking and dragging vertically in the corresponding Amount box.**

! **Note that the Amount range is +/-100. This means that the Amount value can exceed the modulated parameter's range. When this happens, the modulated parameter simply stays at its extreme value until the Amount value gets within the parameter's range again.**

3. **Select the first Destination parameter by click-holding the grey arrow symbol to the right of the corresponding Destination box.**
 The arrow symbol turns blue.

4. **While click-holding, drag to the desired destination parameter on the panel:**



Assigning LFO 1 Rate as a Destination for Envelope 1.

As you hover over a valid destination control on the panel, the parameter name is automatically displayed in the Destination box in the Modulation Bus.

5. **To assign the currently selected Destination control, release the mouse button.**

! **Dragging to the Waveform Display (see "The Sample section") will always assign the playback Position parameter. To assign the sample Start Position or End Position, select the parameter from the list (see below).**

→ **Alternatively, click the desired Destination box and select the Destination parameter from the list.**

The following parameters can be used as modulation Destinations:

Parameter	Description
Position	This affects the sample "playhead" position in the original sample.
Speed	This affects the Speed control in the sample window.
Jitter	This affects the Jitter control in the sample window.
Start Position	This affects the Sample Start marker position in the original sample.
End Position	This affects the Sample End marker position in the original sample.
Pitch	This affects the pitch of the original sample.
Octave	This affects the Oct control in the Playback Algorithm section.
Formant	This affects the Formant Form control in the Playback Algorithm section (if applicable).

Parameter	Description
Sample Level	This affects the Sample Level control of the Playback Algorithm section.
Grain: Length	This affects the Grain Length parameter on the Grain Oscillator and Long Grains playback algorithms.
Grain: Rate/Spacing	This affects the Grain Spacing parameter on the Grain Oscillator and the Rate parameter on the Long Grains playback algorithms.
Grain: X-Fade	This affects the X-Fade parameter on the Long Grains playback algorithm.
Grain: Pan Spread	This affects the Pan Spread parameter on the Grain Oscillator and Long Grains playback algorithms
Grain: Pitch Jitter	This affects the Pitch Jitter parameter on the Grain Oscillator and Long Grains playback algorithms
Spectral Grain: Harm Snap	This affects the Snap parameter on the Spectral Grains playback algorithm.
Spectral Grain: Harm Filter	This affects the Filter parameter on the Spectral Grains playback algorithm.
Spectral Grain: Curve Amount	This affects the Amount parameter on the Spectral Grains playback algorithm.
Oscillator: Modulation	This affects the Mod parameter on the Oscillator.
Oscillator: Octave	This affects the Oct parameter on the Oscillator.
Oscillator: Pitch	This affects the (full range) pitch of the Oscillator.
Oscillator: Level	This affects the Osc Level parameter of the Oscillator section.
Filter: Freq	This affects the Frequency parameter in the Filter section.
Filter: Reso	This affects the Resonance parameter in the Filter section.
Amplifier: Gain	This affects the Gain parameter of the Amplifier section.
Amplifier: Pan	This affects the Pan parameter of the Amplifier section.
Amp Envelope: Attack	This affects the Attack time of the Envelope in the Amplifier section.
Amp Envelope: Decay	This affects the Decay time of the Envelope in the Amplifier section.
Amp Envelope: Sustain	This affects the Sustain level of the Envelope in the Amplifier section.
Amp Envelope: Release	This affects the Release time of the Envelope in the Amplifier section.
LFO's: LFO1/2/3 Delay	This affects the LFO1/2/3 Delay parameter.
LFO's: LFO1/2/3 Rate	This affects the LFO1/2/3 Rate parameter.
Envelopes: Env1/2/3/4 Rate	This affects the Envelope 1/2/3/4 Rate parameter.
Portamento	This affects the Portamento Time parameter.
CV Outputs: CV1/2/3/4 Out	This sends out the source modulation value(s) on the CV1/2/3/4 Output on the rear panel.
Reverb: Decay	This affects the Decay parameter in the Reverb effect.
Reverb: Amount	This affects the Amount parameter in the Reverb effect.
Delay: Time	This affects the Time parameter in the Delay effect.
Delay: Feedback	This affects the FB parameter in the Delay effect.
Delay: Amount	This affects the Amount parameter in the Delay effect.
Delay: Pan	This affects the Pan parameter in the Delay effect.
Dist: Drive	This affects the Drive parameter in the Dist effect.
Dist: Tone	This affects the Tone parameter in the Dist effect.
Dist: Amount	This affects the Amount parameter in the Dist effect.
Compressor: Release	This affects the Release parameter in the Compressor effect.
Compressor: Ratio	This affects the Ratio parameter in the Compressor effect.
Chorus/Flanger/Phaser: Frequency	This affects the center frequency of the Chorus/Flanger/Phaser effects.
Chorus/Flanger/Phaser: Amount	This affects the Amount parameter of the Chorus/Flanger/Phaser effects.
Par EQ: Frequency	This affects the Freq parameter in the EQ effect.
Par EQ: Gain	This affects the Gain parameter in the EQ effect.
Note Trig	This is mainly intended for use with the Display > Mouse Gate or Display > Y-Position source parameters, e.g. with the Freeze motion mode, to trig/gate the sample without having to play the keyboard. This way you can play back the Root Key of the sample by just clicking/dragging in the Waveform display.

Modulation Bus Destination parameters.



6. **Set the Amount for the second Destination (if desired) by turning the corresponding Amount knob, or by clicking and dragging vertically in the Amount box for the second destination.**
7. **If desired, select a second Destination parameter by click-holding the blue arrow symbol to the right of the corresponding Destination box, and dragging to the desired control on the panel.**
8. **If desired, click the Scale box and select a Scale parameter.**
 The available Scale parameters are the same as the Source parameters, see "Modulation Bus Source parameters.".
9. **Turn the Scale Amount knob, or click the Amount box to the left of the Scale box and move the mouse pointer up or down to set a Scale Amount value.**
 Both positive and negative Scale Amount values can be set (+/- 100%). If you, for example, are using the Mod Wheel as Scale parameter and don't want any modulation when the Mod Wheel is set to zero, set the Scale Amount parameter to 100%. Then, there will be no effect when the Mod wheel is set to zero, and full modulation when the Mod Wheel is all the way up.

- **How much modulation will be applied when the Scale parameter is set to maximum is governed by the to Destination Amount parameter(s).**
- **How much the Scale parameter controls the modulation is set with the Scale Amount parameter.**

→ **To clear an assigned Source, Destination or Scale parameter, hold down [Ctrl](Win) or [Cmd](Mac) and click the Source/Destination/Scale box. Alternatively, click the Source/Destination/Scale box and select "Off" from the list.**

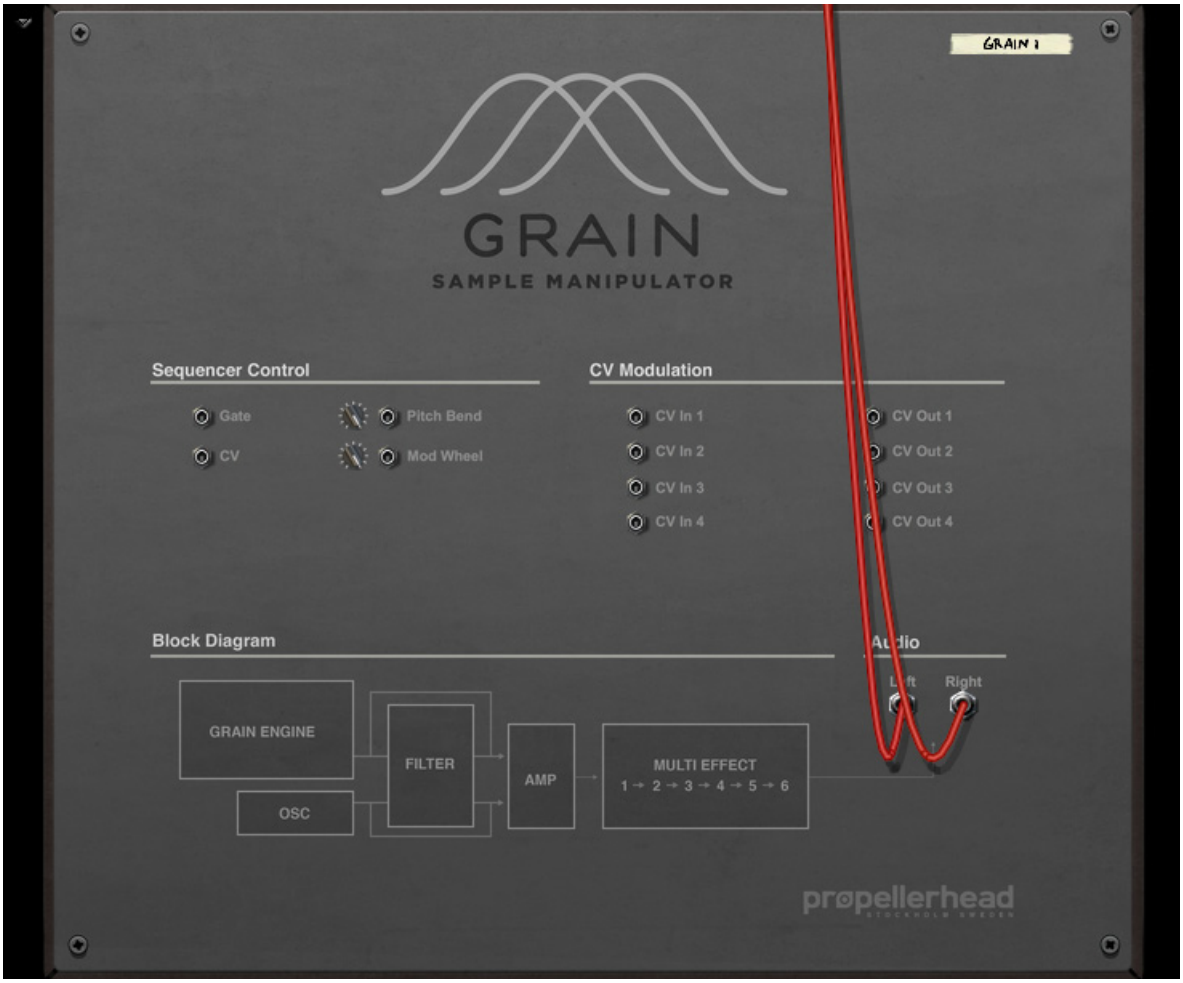
→ **To reset an Amount value to 0, hold down [Ctrl](Win) or [Cmd](Mac) and click the desired Amount box or knob.**

→ **To clear an entire modulation assignment (a whole row), click the circular X button to the right of the corresponding Scale box.**

Modulation example patches

The factory patches for Grain features a folder named "Templates". In this folder you will find a number of patches that show typical modulation examples, to make it easier to get the hang of how to create your own patches.

Connections



! Remember that CV connections are NOT stored in the Grain patches! If you want to store CV connections between devices, put them in a Combinator device and save the Combi patch.

Sequencer Control inputs

The Sequencer Control CV and Gate inputs allow you to play Grain from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity. There are also inputs for modulating the Pitch Bend and Mod Wheel parameters.

CV Modulation inputs and outputs

These assignable control voltage (CV) inputs and outputs can be used for modulation of and from assigned Source and Destination parameters in the Modulation Bus section, see "The Modulation Bus section".

Audio Output

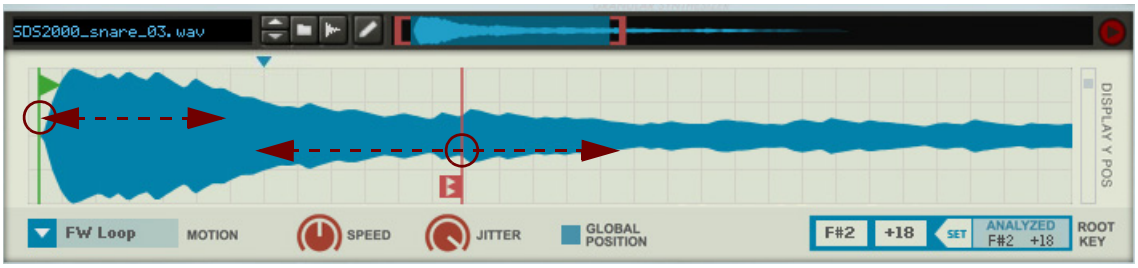
These are the main audio outputs. When you create a new Grain device, these outputs are auto-routed to the first available outputs in the I/O device.

Tips and Tricks

Automating sample playback parameters from the sequencer

Besides the extensive modulation capabilities of the Modulation Bus, the sample playback parameters can also be automated in the main sequencer. For example, you could automate the Sample Start and/or Sample End markers in the Sample section to have the markers reposition in real-time during playback of the main sequencer. Below is a basic example of how you could automate various sample playback parameters:

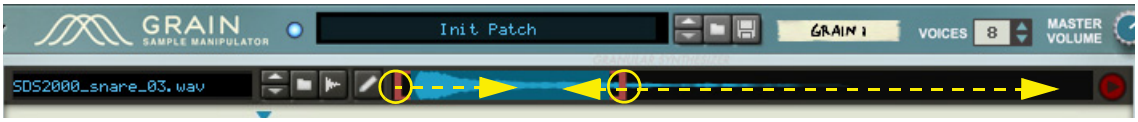
1. **Record some notes on the Grain sequencer track in the main sequencer and then hit Stop twice.**
2. **Hit Record again in the main sequencer and drag the Sample Start and Sample End markers during recording:**



3. **Hit Stop in the sequencer twice when you are done recording.**
 If you moved both the Sample Start marker and the Sample End marker you will now have four parameter automation lanes/tracks in the sequencer.

- **The Display Y lane/track and the Display Gate lane/track always appear as soon as you have recorded any marker movements in the Sample section display.**
- **The Display Y lane/track represents the vertical movements you made with the mouse during the recording.** This automation doesn't affect the sample playback in any way but can instead be used as a modulation source in the Modulation Bus, for modulating the desired destination parameter(s).
- **The Display Gate lane/track reflects when you clicked (and held) the mouse button during the recording.** This automation doesn't affect the sample playback either but can be used as a modulation source in the Modulation Bus, for modulating/gating the desired destination parameter(s).
- **The Position and End Pos lanes/track represent the movements of the Sample Start and Sample End markers respectively.**

→ **If you like you could also record automation of the Sample Range Zoom and Scroll parameters by dragging the markers in the Sample Overview area:**



After recording the movements of the Sample Range markers, two new Parameter Automation lanes/tracks appear:

- **The Scroll lane/track represents the movement of the leftmost Sample Range marker, and thus the movement of the entire sample range.**
- **The Zoom lane/track represents the movement of the rightmost Sample Range marker, and thus the zooming (in or/and out) of the sample range.**

► **It's also possible to automate the Motion, Speed, Jitter and Global Position parameters.**

► **Also, don't forget to check out the modulation example patches, see "Modulation example patches".**

Chapter 15
Mimic Creative Sampler

Introduction



The Mimic Creative Sampler is a powerful yet very straight-forward sampler, tailor-made for quick and easy triggering, chopping and manipulation of samples. It features eight sample slots, where each slot can hold one sample. Each slot also has its own complete synth parameters setup, with pitch controls, filter, envelopes, LFO and effects.

You could either load a sample from your computer or sample straight into Mimic (Reason stand-alone only). You can then select various sample playback modes and high-quality stretch algorithms to manipulate and process the audio. You could also use Mimic as a traditional sampler and just play back samples in a regular “tape-style” fashion.

Mimic also features a great-sounding “lo-fi style” multi-effect to spice up your sounds even more.

Don’t forget to check out the Mimic video tutorial here!

Panel overview

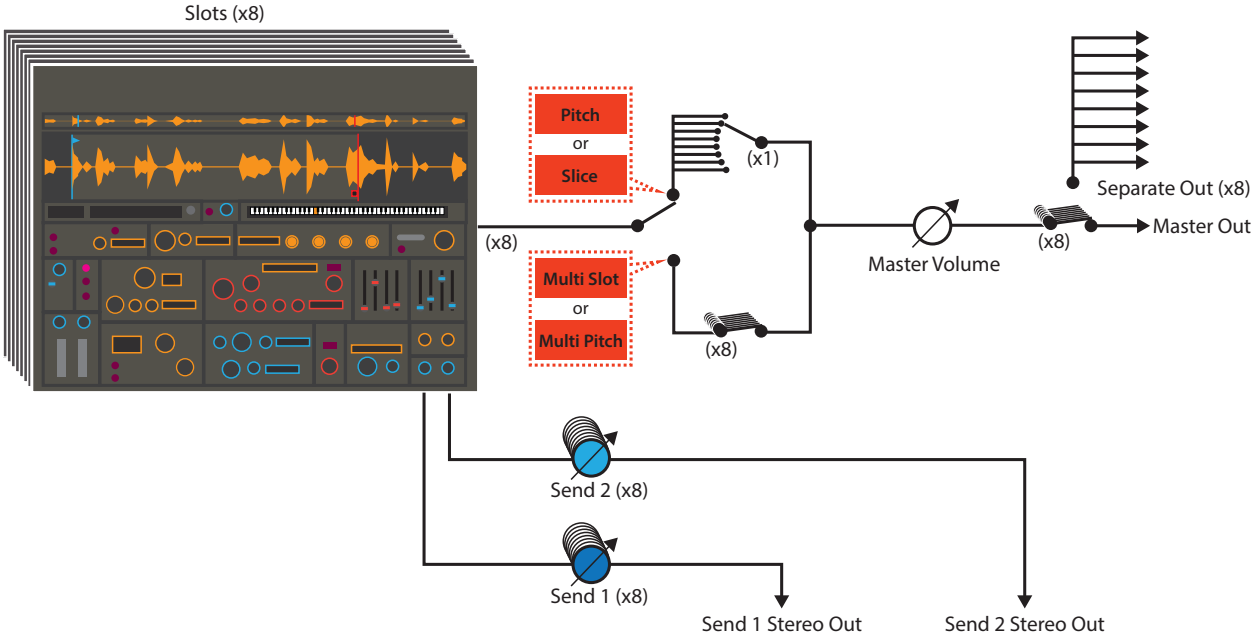
The Mimic front panel contains the following sections:



The Mimic front panel sections.

- **1. MIDI Note On LED.**
- **2. Patch selector (for browsing, loading and saving patches).**
- **3. Master Volume controls.**
- **4. Playback Mode section.**
- **5. Slot selector.**
- **6. Sample section (for sample loading/sampling and sample playback functions).**
- **7. Playback Start Position, Speed, Stretch and Slices sections.**
- **8. Performance and “play” controls.**
- **9. Pitch section.**
- **10. Filter section.**
- **11. Envelopes section.**
- **12. LFO section.**
- **13. Amp section.**
- **14. Compressor, Effects and EQ sections.**

Signal flow



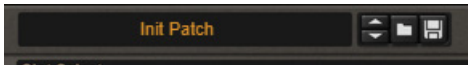
Mimic signal flowchart.

- **Mimic features eight Slots, where each Slot holds a complete configuration of sample parameters and panel parameter settings.**
 Each Slot features a Sample section, where you can load your sample - or sample your own (Reason stand-alone only). You can then select if you want to play back the sample as Slices or Pitched (chromatically). You can also choose among various sample Stretch algorithms. The signal is then routed through a complete configuration of synth parameters, such as Pitch, Filter, Amp and Effects.
- **Depending on the selected Slot Mode, the signals from the Slots are be routed a bit differently.**
 In Pitch Mode and Slice Mode, only one single Slot can be played back at a time.

 In Multi Slot Mode and Multi Pitch Mode you can play back up to eight Slots simultaneously.
- **The synth sections in Mimic's eight Slots are described in "Panel reference".**

Playing and using Mimic

Loading and saving patches



A Mimic patch contains the parameter settings for all used Slots, i.e. up to eight complete parameter setups. Loading and saving patches is done in the same way as with any other internal Reason device, see "Loading patches" and "Saving patches" for details.

! **Like with the other sampler devices in Reason, the patch does not include the actual sample - only a reference to it. Therefore, the sample has to be stored separately (self-contained with the song, or already on disk or in a ReFill on your computer).**

Global output controls



Master Volume

This is the main stereo output volume control and controls the volume of all eight Slots together.

Slot Mode



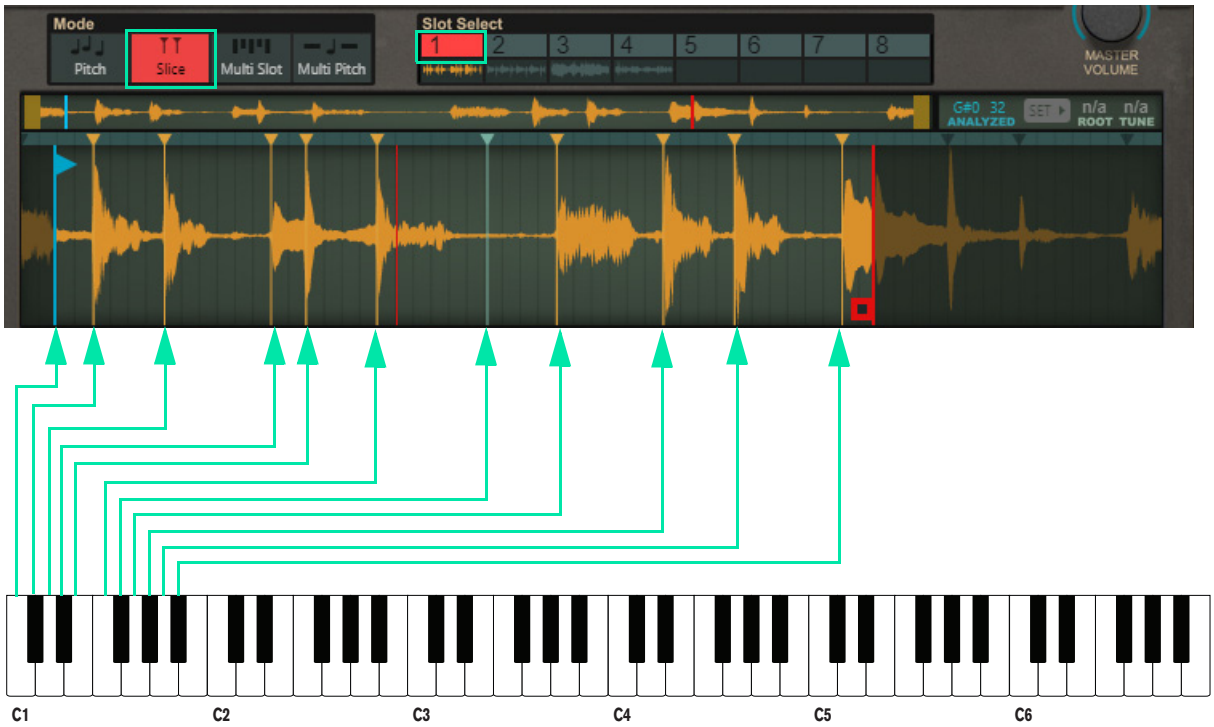
Mimic has four Slot Modes, which defines how the sample of each Slot should play back - and also if you can use one or several Slots simultaneously:

Pitch Mode



In Pitch Mode you select one of the eight Slots for melodic playback (playing the sample from the start to the end locators). The sample can then be played chromatically pitched over the entire keyboard range.

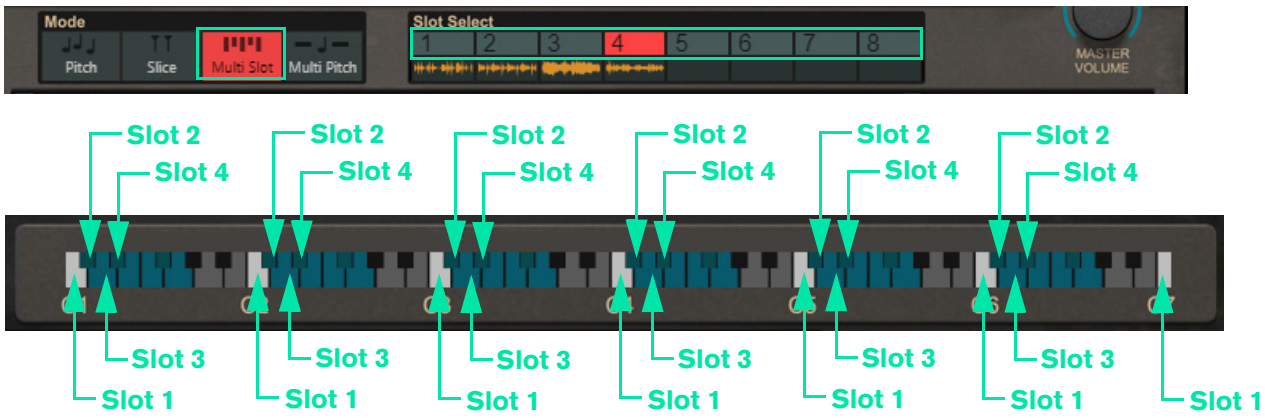
Slice Mode



In Slice Mode you select one of the eight Slots for playback, and slice a (longer) sample, manually and/or automatically. The slices are then triggered from the keyboard (chromatically, with the leftmost slice playing back from note C1) without affecting the original pitch of the sample. See "Slices" for more details.

Multi Slot Mode

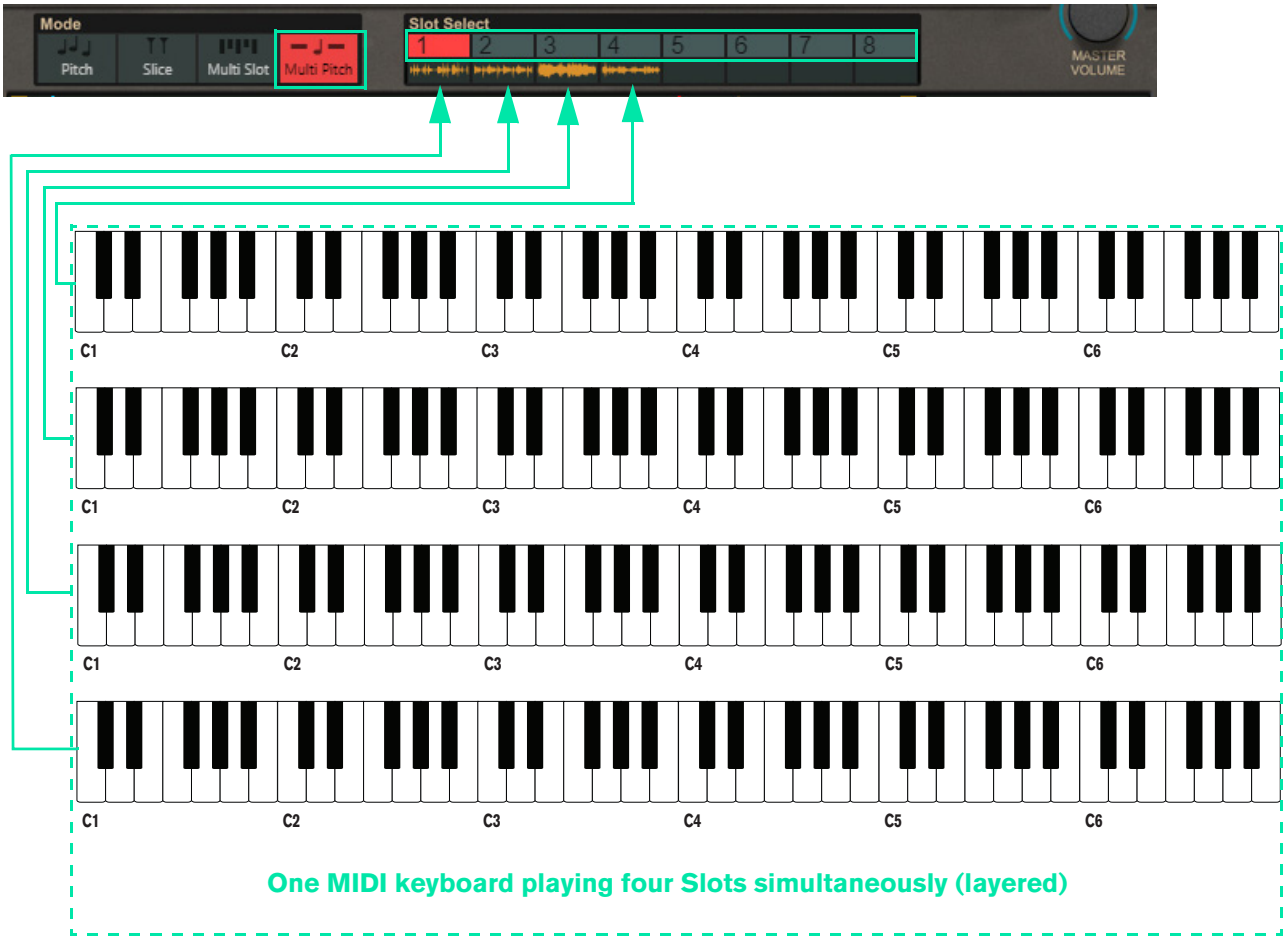
In Multi Slot Mode different keys (C-G in each octave) triggers Slots 1-8 (from the start to the end locators), drum-machine style - without affecting the sample pitches. This is displayed in the Note/range indicator as follows (the currently selected Slot is high-lighted on the keyboard - the others are dark blue):



Multi Slot Mode with samples loaded in Slots 1-4.

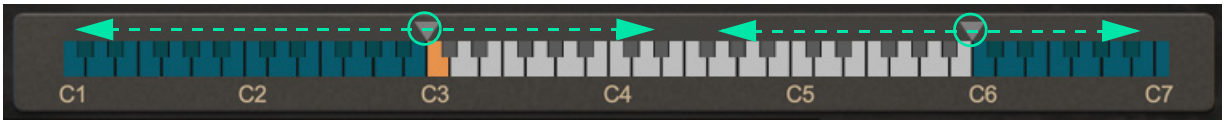
► **Multi Slot Mode works great with pad controllers.**

Multi Pitch Mode



Multi Pitch Mode with samples loaded in Slots 1-4, played back in a layered fashion.

In Multi Pitch Mode you can play back up to eight samples (one sample per Slot) simultaneously, for melodic playback (playing the sample from the start to the end locators). You can set different keyboard ranges (overlapping and/or adjacent) for the eight Slots and then play back the samples chromatically pitched. You can set the desired keyboard range in the Note/range indicator as follows (the currently selected Slot is high-lighted on the keyboard - the others are dark blue):



Setting the key range for the currently selected Slot.

1. **Click the desired Slot button (see "Slot Select" below) to select it.**
2. **Drag the left and right note range markers to set the desired note range.**
 The minimum "range" for a Slot is one note.
3. **Click another Slot button and set a new desired note range.**

► **Note that Slots can have overlapping note ranges (also completely overlapping so that you could play all eight Slots together in a layered fashion).**

Slot Select



Mimic has eight Slots, which can hold one sample each. Each Slot also has a complete parameter setup on the Mimic front panel, so each slot can have its own unique parameter configuration.

→ **Click the desired Slot Select button to bring up all parameters for that slot, including the sample and its settings.**

! **Note that in Pitch Mode and Slice Mode (see above), only one single Slot can play back its sample at a time (as indicated by the grayed out waveforms below the other Slot Select buttons).**

Performance and "play" controls

Each of the eight Slots has their own individual set of performance/play controls on the front panel:



Porta

Portamento makes note pitches glide from previous notes to new ones, at the time set with the Time knob. Portamento can be used in all Key modes (see above).

- **When On in Poly Key Mode (see below), the pitches will glide from any of the available voices.**
 The results will be unpredictable since there is no way of controlling from which note(s) the glide(s) will commence. The effect is very nice, though.
- **When On in Mono Retrig or Mono Legato Key Mode (see below), the pitch will glide between consecutive notes.**
- **In Auto mode, the pitch will glide between consecutive monophonic notes only when you play legato. If you have selected Poly Key Mode (see below), Auto will have no effect at all.**
 If you release the previous key before hitting the new key, there will be no portamento effect.

Key Mode

Here you choose how Mimic should respond to MIDI Note data:

- **Poly**
 Select this if you want to play Mimic polyphonically.
- **Mono Retrig**
 Select this if you want to play Mimic in monophonic mode and always retrigger the envelopes as soon as you play a new note.

- **Mono Legato**
 The Mono Legato mode is also monophonic. However, if you play a new note without having released the previous one, the envelopes and sample playback position won't start over.

► **Also see the description of the "Global Position", which allows new notes to continue playing at the current sample playback position (and not restart playback).**

Pitch Range

→ **Set the desired Pitch Bend range for the "Pitch Bend" wheel with the up/down buttons, or by click-holding on the display and dragging up/down.**
Range: +/-24 semitones (+/-2 octaves) in steps of +/-1 semitone.

Pitch Bend

The Pitch Bend wheel can be used for bending note pitches up and down. Mimic also responds to Pitch Bend MIDI data from a connected MIDI master keyboard. You set the desired Pitch Bend Range with the "Pitch Range" control above the Pitch Bend wheel.

! **Note that if you have selected the Tape Stretch algorithm (see "Tape"), pitch bending the sample will also affect its playback speed.**

Mod Wheel

The Mod Wheel can be used for modulating a number of parameters in the eight Slots of Mimic. Use the Mod Wheel as a source parameter for the panel parameters that feature modulation source drop-down selectors.

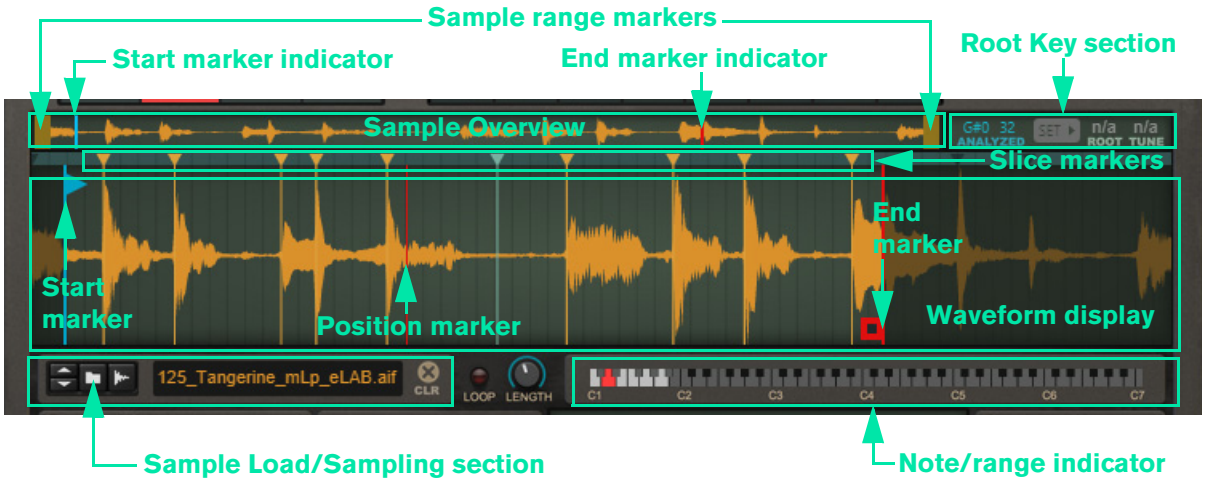
LFO Scale

The LFO Scale knob can be used for scaling the LFO amount with the Mod Wheel - perfect for gradually introducing Vibrato effects, for example. If the LFO Scale knob is set to 100% the LFO Amount will be 0 when the Mod Wheel is at 0. See "The LFO section" for more details about the LFO.

Panel reference

The Sample section

Here is where you load/sample the audio for the currently selected Slot. The Waveform display differs somewhat, depending on which Slot Mode you have selected (see "Slot Mode").



A sample loaded in a Slot with Slice Mode selected.

Loading samples

- **Load a sample using drag & and drop, or by clicking the Browse sample button, or by using the Up/Down buttons to scroll and load a sample from the currently selected folder.**
- **Drag a sample from the browser and drop on a Slot Select button, to load the sample in the desired Slot.**

! **It's possible to load/sample stereo signals. However, the waveform will always be displayed as a mono signal, regardless if it's mono or stereo.**

Root Key

A sample is automatically analyzed for its original pitch at the Sample Start position. The analyzed pitch is displayed in the Root Key section. If you move the Sample Start marker, the sample is automatically re-analyzed.

- **Click the "SET" button to use the analyzed Root Key.**
 This will automatically place the analyzed Root Key on the correct note on the keyboard, as indicated by an orange key in the Note/range indicator below the Waveform display.
- **Alternatively, set the Root Key manually by dragging up/down in the "Root" and "Tune" (cent) boxes:**

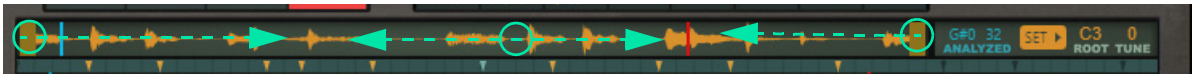


! **Note that the Root Key function only works in Pitch Mode and Multi Pitch Mode.**

Setting the sample range

First you could decide how much of the original sample you want to use - and where in the sample you want to work:

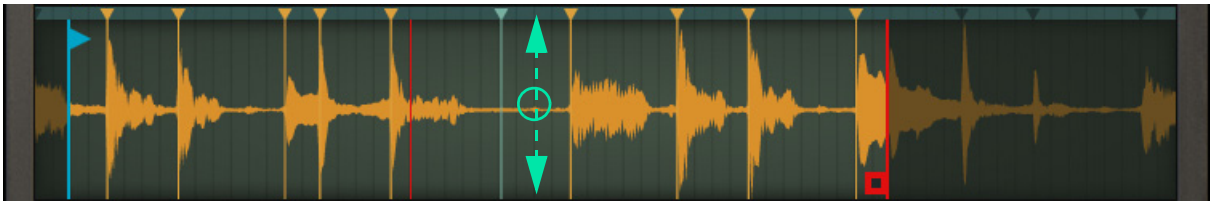
→ **Zoom and/or scroll in the Sample Overview to define the Sample range you want to work in.**



To scroll, click and drag sideways between the dark yellow sample range markers. To zoom, click and drag any of the sample range markers sideways. The set Sample range is automatically updated and displayed in the waveform display.

→ **To work in the entire Sample range, drag the left Sample range marker all the way to the left, and the right Sample range marker all the way to the right, in the Sample Overview.**

Zooming in the Waveform Display



→ **Click and hold the mouse button anywhere in the waveform display, then drag down to zoom in, and up to zoom out in the waveform.**

← **Dragging sideways in the Waveform Display will scroll left/right.**

Setting the sample start and end



→ **Drag the Start and End marker handles to where you want the sample to begin and stop playing back.**
Note that the Start and End markers cannot be set in reversed order. If you want to play the sample backwards, from the End marker to the Start marker, click the Reverse button (see "Reverse").

• **If you drag and move the Start/End marker up into the Slice Marker area above the waveform, the Start/End marker will snap to the closest slices. This works in all modes, not just in Slice Mode:**



→ **Alternatively, double click in the Waveform Display to position the Start marker. Hold down [Shift] and double click to position the End marker.**

Previewing samples

→ **Preview the sample by holding down [Option]/[Alt] and clicking at the desired position in the Waveform Display.**
The sample will play back for as long as the mouse button is depressed.

– **It's possible to automate the sample start and end settings, see "Automating the sample Start and End markers".**

Loop and Loop Length



It's also possible to loop (Forward Loop) samples - or slices in Slice Mode. The loop always happens at the end of the sample/slice. In all modes except for Slice Mode the Loop Length is visually indicated with a transparent red region in the Waveform Display, so you can see exactly where the loop is.

→ **Enable Loop by clicking the Loop LED.**

→ **Adjust the Loop Length by turning the Length knob, or by holding down [Command]/[Ctrl] and clicking/dragging sideways in the waveform display:**



If you are playing back the sample reversed (see "Reverse") the looped region originates from the Start Marker instead and is displayed in transparent blue.

! **In Slice Mode the looped region is not indicated visually in the Waveform Display.**

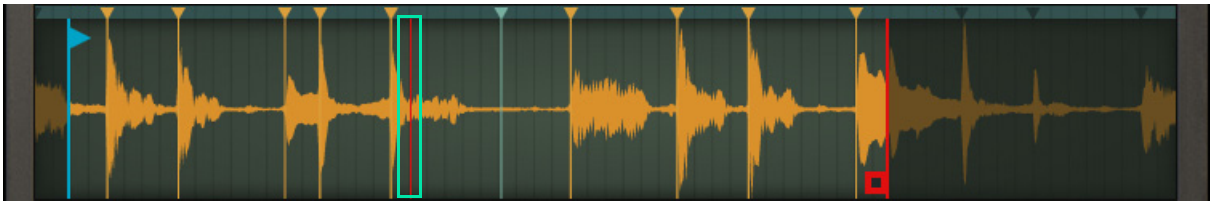
Start Position



The Start Position section contains controls for determining where the sample/slice playback should begin, and in which direction the sample/slices should be played back.

Global Position

→ **Click the Global Position LED to start the playback of new/additional voices from the global position, i.e. from where the red Position marker is currently positioned in the Waveform Display:**



This function is great for polyphonic rhythmic sounds, where you want to have all the voices synced in time.

If not active, new/additional voices will always start playing back from the blue Sample Start marker.

Reverse

→ **Click the Reverse LED to have the sample/slices play back backwards, from the end to the start.**

Start Position Mod

→ **Turn the Mod knob to set a modulation amount for the sample/slice playback start position.**
Note that the Mod amount is bipolar (-/+) so that you can modulate the start position either earlier or later than the default start position.

→ **Select a Modulation source from the drop-down selector, to modulate the sample/slice start position.**

! **The Start Position Modulation behavior also depends on the "Snap to Slices" setting described below.**

Snap to Slices

→ **Click the Snap to Slices LED to always start the playback from a Slice Marker (and not from somewhere in between Slice Markers).**

! **Note that this function is only useful in when modulating the Start Position (see above) in Pitch Mode, Multi Slot Mode and Multi Pitch Mode, since the Start Position always snaps to slices in Slice Mode (see "Slice Mode").**

Speed and Speed Mod



The Speed control determines how fast the play position ("playhead") moves in the waveform.

Speed

→ **Set the sample playback speed with the Speed knob.**
Depending on which Stretch type (see "Stretch") is currently selected, the sonic result will vary. If you have selected the Tape Stretch type (see "Tape"), the Speed knob also affects the pitch. Note that the Speed can be set all the way down to 0%, i.e. "stop". Great for Tape Stop effects in Tape mode and for static playback in other Stretch types, for example.

Speed Mod

→ **Modulate the Speed with the Mod knob.**

→ **Assign a modulation source from the drop-down selector next to the Mod knob.**
For example, selecting an Envelope as a modulation source could cause the playback speed to start fast and gradually slow down when you hold a note - or vice versa.

Stretch



Mimic features five different Stretch types, which can be selected from the drop-down selector. Depending on the selected Stretch type, there are also some additional controls to modify the sample characteristics.

Tape



This is the good old "tape recorder" type, where speed and pitch are coupled. This means that to achieve a higher pitch you simply increase the playback speed of the sample/slices - and vice versa.

With the Tape stretch type selected, there is a Loop X-Fade knob present. This controls the crossfade amount when the Loop function is active for the sample/slices (see "Loop and Loop Length").

If Loop is off, the Loop X-Fade knob has no effect.

Advanced



This is a high-quality stretch algorithm suitable for most type of polyphonic and complex audio material.

With the Advanced stretch type selected, there is a Preserve Transients button present. Transients are regions in the sample where the level quickly goes from quiet to loud, for example in percussive hits and other types of "attacks".

→ **Click the Preserve Transients button to preserve any transients in the sample/slices.**
When off, any transients will be "smeared out" and less prominent, which might be desired in some situations.

Melody



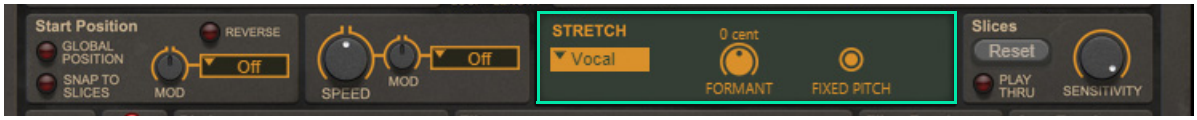
This is the Melody stretch type used for audio in the Reason sequencer, i.e. a high-quality stretch algorithm suitable for monophonic audio material.

With the Melody stretch type selected, there is a Preserve Transients button present.

→ **Click the Preserve Transients button to preserve any transients in the sample/slices.**
When off, any transients will be "smeared out" and less prominent, which might be desired in some situations.

← **The Melody stretch type is well suited if you are using the Loop function (see "Loop and Loop Length"), since it usually reduces clicks/pops at the loop point.**

Vocal



This is the Vocal stretch type used for audio in the Reason sequencer, i.e. a high-quality stretch algorithm suitable for monophonic vocal audio material.

With the Vocal stretch type selected, there are two additional controls present:

- **Turn the Formant knob to change the formant of the sample/slices.**
 Turning this up will be like creating a smaller "body" for the sound, and turning it down will be like creating a larger body. If you are using a vocal sample, changing the Formant would be like changing the character from "adult" to "child" like.
- **Click the Fixed Pitch button to "auto-tune" the sample/slices to the currently played (note) pitch.**
 This is really cool for creating processed vocals that you could pitch from the keyboard.

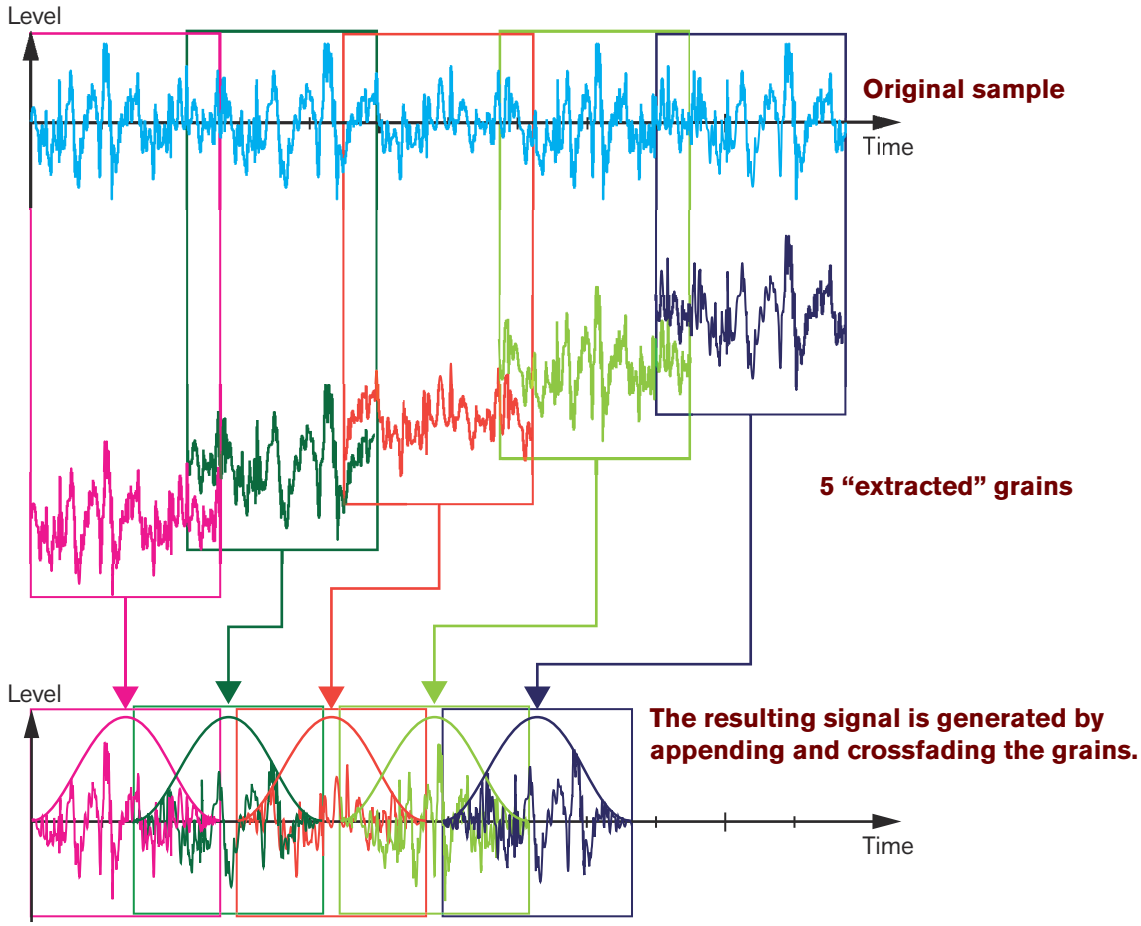
Granular

This is a "vintage" type of digital pitch shift/stretch method, where grains of the sample are being looped and cross-faded.



The Granular stretch type utilizes playback of a series of snippets of audio data - grains - "extracted" from the sample. The grains could be of a selectable length and overlap. The grains could then be played back in a number different ways - with or without crossfades between the grains.

The picture below shows the basic principle of the Granular Stretch type:



An example of a signal generated from 5 grains of a sample.

Here is what happens in the example above:

- **The original sample at the top is used as base for the granular stretch.**
- **5 grains (of the same lengths and the same distances between them) are “extracted” from the original sample.** The distance between the grains is determined by the current sample playback speed. The grains could contain common audio data in some parts (like in the beginnings and ends in the example above).
- **The 5 grains are then placed after one another, partly overlapping each other.** The distance between the grains is determined by the playback rate.
- **When the grains are played back, big parts of the grains are played back together (since they are overlapping).** In the example above, there are also crossfades between the grains to make the overlaps smoother.

With the Granular stretch type selected, there are four additional controls present:

- **Control the grain lengths with the Length knob.** This sets the lengths of the grains. At 0% you get the shortest grains and the sound could be a little gritty at this setting. At 100% you get the longest grains.
- **Control how much the grains should overlap each other with the Overlap knob.** This sets how much the grains should overlap each other and also the crossfade amount. At 0% you get no overlap and almost no crossfade at all. At 100% you get the longest overlaps and also smooth crossfades between the grains.

→ **Set the playback position deviation with the Jitter knob.**
The Jitter function modulates the sample playback position minutely and randomly. The Jitter function can be great for generating "chorus"-like effects and to make a sound more "alive", depending on the other settings in the sound. At 0%, the timing and playback position is completely accurate and at 100% it is completely random.

→ **Set how much you want the grains to be panned in the stereo panorama with the Spread knob.**
0% means the signal will be unaffected and 100% means every other grain will be panned hard left and hard right. Great for nice and wide stereo effects!

Slices



This section is mainly useful when you are working in Slice Mode (see "Slice Mode").

← **Even though slices are the core of Slice mode, they are available in the other modes too. They can be used for snapping the Start position (see "Setting the sample start and end") - either when you drag it manually or when you modulate or automate it.**

Sensitivity

Slices are added automatically at transients according to the Sensitivity knob setting.

→ **Turn the Sensitivity knob to increase or decrease the number of automatically detected slices.**
Automatically generated slice markers are indicated in yellow.

• **In Slice Mode the number of available keys changes according to the number of slices in the loop. For example, if there are 10 slices in the loop, the first 10 notes (counted upwards from C1) are used. If the loop contains 15 slices, the first 15 notes are used, and so on:**



! **Note that the maximum number of slices that can be detected and used is 92 (from C1 to G8). If you are using a very long sample and a high Sensitivity setting, the detected slices might not cover the entire sample - only the first part. If you want to access slices further into the sample you might therefore have to move the Sample Start marker (or reduce the Sensitivity setting).**

Moving, adding and removing Slice markers

→ **Drag the Slice markers sideways to move them:**



→ **Add slices manually by double clicking in the Slice Marker lane directly above the waveform:**



! **Any manually moved/added slice markers are automatically indicated in a different color, which also means they are no longer affected by the Sensitivity knob.**

► **You can also click on a slice marker handle (without moving it) to deactivate it from the Sensitivity knob.**

→ **Delete slice markers by double clicking the slice marker handles in the Slice Marker lane.**

Reset

→ **Click the Reset button to restore the slices to the ones auto-generated by transient detection/Sensitivity.**
Any manually added slice markers will be removed.

Play Thru

→ **Click the Play Thru LED to force the playback to continue beyond the following slice markers, for as long as the notes sustain.**
When Off, the playback will automatically stop at the next slice marker, even if you have sustaining notes.

Pitch



Semi

→ **Set the pitch in semitone steps.**
Range: +/-24 semitones (+/-2 octaves).

! **Note that if you are using the Tape Stretch type (see "Tape"), the pitch settings also affect the playback speed.**

Tune

→ **Changes the pitch in steps of 1 cent.**
Range: +/- 50 cents (down or up half a semitone).

LFO

→ **Sets how much the pitch should be affected by the LFO (see "The LFO section").**
Range: +/- 100%.

← **By using the LFO Scale function (see "LFO Scale"), you can gradually introduce the LFO modulation amount by using the Mod Wheel.**

Pitch Mod

→ **Sets how much the pitch should be modulated by the source assigned in the drop-down selector to the right.**
Range: +/- 100%.

The Filter section

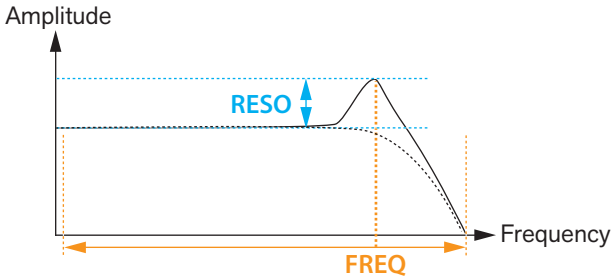


The Filter section features eight different filter types.

Filter Type selector

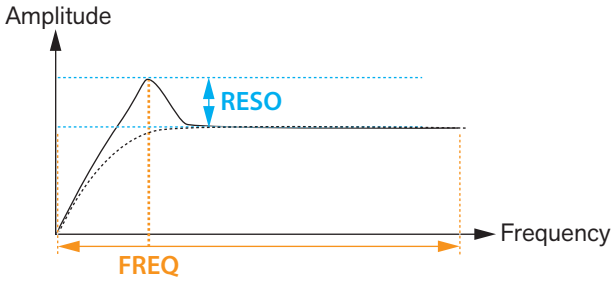
→ **Click the Filter type drop-down selector to choose any of the following filter types:**

- **LP 24**



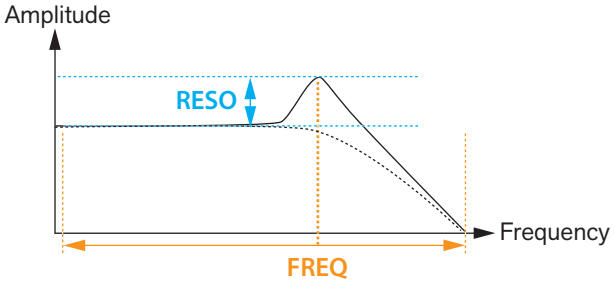
This is a standard 24dB/octave lowpass filter. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob.

- **HP 24**



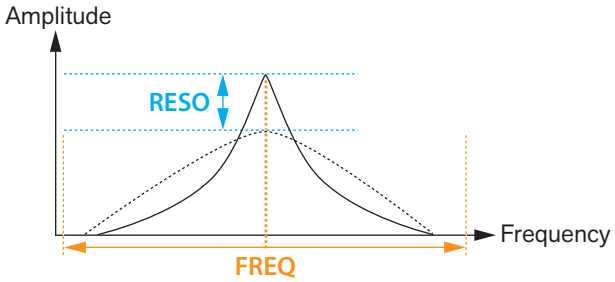
This is a standard 24dB/octave highpass filter. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob.

- **LP 12**



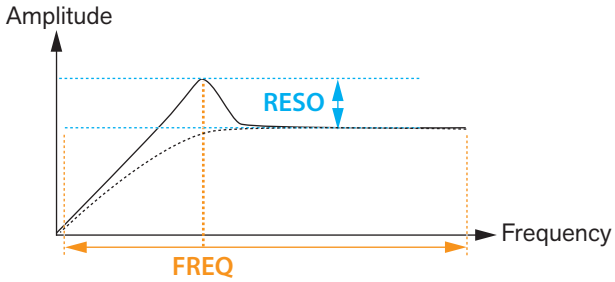
This is a standard 12dB/octave lowpass filter. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob.

- **BP 12**



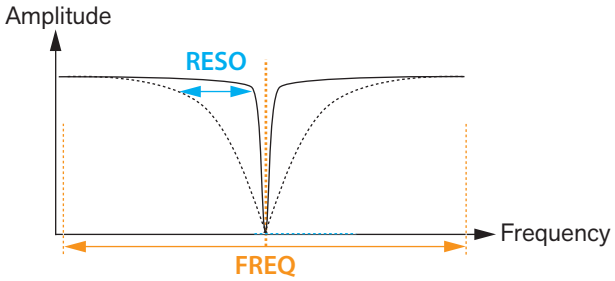
This is a standard 12dB/octave bandpass filter. Set the center frequency with the Freq knob and the resonance amount with the Reso knob. Note that raising the resonance also makes the passband narrower.

- **HP 12**



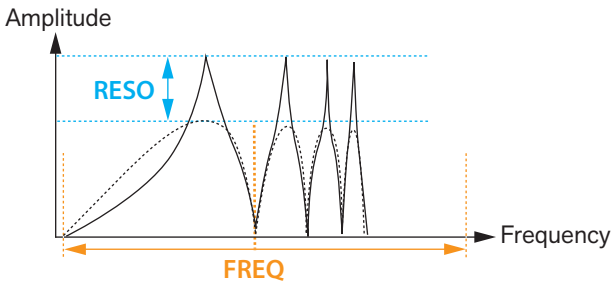
This is a standard 12dB/octave highpass filter. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob.

- **Notch**



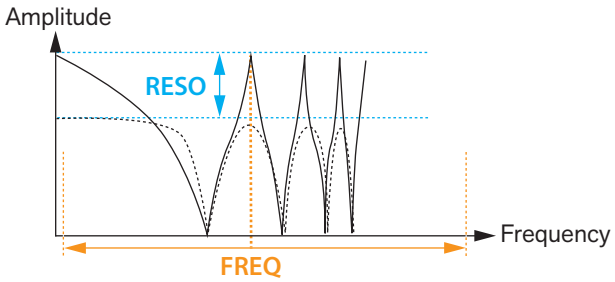
A notch filter (or band reject filter) could be described as the opposite of a bandpass filter. It cuts off frequencies in a narrow midrange band, letting the frequencies below and above through. Set the center frequency with the Freq knob and the notch width with the Reso knob. The higher the Resonance, the narrower the notch.

- **Comb -**



This is a comb filter with a positive feedback loop - but without feed forward - ideal for flanger and phaser types of effects. Set the cutoff frequency with the Freq knob and the resonance amount with the Reso knob. The difference between “Comb +” (see below) and “Comb –” is in the position of the peaks in the spectrum. The main audible difference is that the “Comb –” version causes a bass cut.

- **Comb +**



This is a multi notch filter, great for phaser types of effects. Set the cutoff frequency with the Freq knob and the attenuation amount - and consequently the bandwidth - of the notches with the Reso knob. The difference between “Comb +” and “Comb –” (see above) is in the position of the peaks in the spectrum. The main audible difference is that the “Comb +” version lets through more bass frequencies.

Freq

→ **Set the cutoff frequency (for the HP and LP filter types) or the center frequency (for the BP and Notch filter type).**
Range: 37.0 Hz to 16.00 kHz.

Reso

- **Set the resonance amount.**

Drive

- **Set the amount of overdrive distortion in the filter.**

Kbd

- **Set how much you want the filter cutoff/center frequency to track the keyboard.**
 At 0%, the filter frequency is static regardless where on the keyboard you play. At 100% the filter tracks the keyboard 1:1, i.e. one semitone per note.

Vel

- **If you want the Filter Envelope amount to be controlled from keyboard velocity, turn up the Vel knob.**

Env

- **Set the cutoff/center frequency modulation amount from the Filter Envelope (see "The Filter Envelope and Amp Envelope sections").**
 Range: +/- 100%.

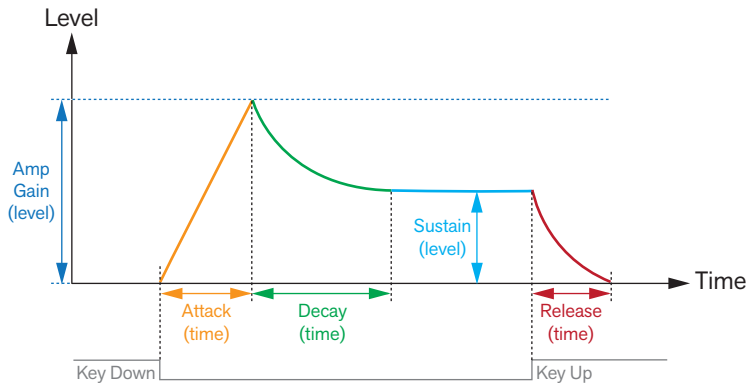
Freq Mod

- **Select a modulation source, for modulating the filter cutoff/center frequency, from the drop-down selector.**
- **Set the desired modulation amount with the Mod knob.**
 Range: +/- 100%.

The Filter Envelope and Amp Envelope sections



The Filter Envelope and Amp Envelope feature standard ADSR envelopes that control the modulation amounts of the respective destinations (the Filter Freq and the Amp Gain). The picture below shows the various stages of the ADSR envelope:



The ADSR envelope stages.

A(ttack)

When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the maximum frequency value (Filter Envelope) or Gain level (Amp Envelope). How long this should take, depends on the Attack setting. If the Attack is set to "0", the maximum Freq/Gain value is reached instantly. If the Attack value is raised, it will take longer time before the maximum Freq/Gain value is reached.

D(ecay)

After the maximum Freq/Gain value has been reached, the level starts to drop. How long this should take is governed by the Decay parameter.

If you want to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0", the Decay parameter should be set to a medium value and the Sustain level should be set to "0", so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you raise the Sustain parameter.

S(ustain)

The Sustain parameter determines the level the envelope should rest at, after the Decay stage. If you set Sustain to full level, the Decay setting is of no importance since the frequency/volume of the sound is never lowered.

If you want to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

But often a combination of Decay and Sustain is used to generate envelopes that rise up to the maximum Freq/Gain value, then gradually decreases to finally land to rest on a level somewhere in-between zero and the maximum Frequency/Gain value. Note that Sustain represents a level, whereas the other envelope parameters represent times.

R(elease)

The Release parameter works just like the Decay parameter, except it determines the time it takes for the Freq/Gain to drop back to zero (or to the set Freq value) after you release the key.

The LFO section



An LFO (Low Frequency Oscillator) is used for generating cyclic modulation. A typical example is to have an LFO modulate the pitch of a signal to produce vibrato, but there are countless other applications for LFOs.

The LFO section features one general purpose LFO, which can be assigned to control selectable parameter(s) in other sections on the front panel.

- **By using the LFO Scale function (see "LFO Scale"), you can gradually introduce the LFO modulation amount by using the Mod Wheel.**

Wave

- **Select an LFO waveform by clicking the spin controls to the right of the waveform display, or by click-holding in the display and dragging up or down.**
 Besides the standard waveforms (sine, triangle, pulse, etc.) there are random, slope and stepped waveforms. The shape of the waveforms are shown in the display.

Rate

- **Set the LFO frequency with the Rate knob.**

Key Sync

- **Click the Key Sync button to restart the LFO at every new Note On.**

Beat Sync

- **Click the Beat Sync button to sync the LFO to the main sequencer Tempo.**
 The Rate parameter now controls the time divisions.

Delay

- **Turn the Delay knob to introduce a delay before the LFO modulation kicks in after a note is played.**
 Turn clockwise for longer delay times.

The Amp section



Gain

→ **Set the desired maximum level for the Amplifier with the Gain knob.**
This is the maximum level the envelope will reach after the Attack stage is completed (see "A(ttack)".).

Vel

→ **If you want the Gain level to be controlled from keyboard velocity, turn up the Vel knob.**

Gain Mod

→ **Assign a modulation source in the drop-down selector to the right. Then, control the modulation amount with the Mod knob.**

► **If you want to create a tremolo effect, select the LFO as Source in the drop-down selector.**

Pan

→ **Set the panning of the output signal from the Amplifier in the stereo panorama.**

Pan Mod

→ **Assign a modulation source in the drop-down selector to the right. Then, control the modulation amount with the Mod knob.**

► **Since Pan works individually per voice, you can assign e.g. Keyboard Velocity, an Envelope or "Random" as source to create cool panning effects.**

The Compressor



This is a Compressor, which can be used for compressing the signal and evening out the signal levels.

→ **Turn the Squeeze knob to set the compression amount.**
The red LED above the knob indicates the signal compression.

The Effect section



The Effect section features seven types of distortion/modulation effects, to spice up your sound.

→ **Select the desired Effect algorithm from the drop-down selector.**
"Noise" adds noise to the signal (when a signal is present).

"Reso Noise" adds resonant noise to the signal (when a signal is present).

"Ring Mod" is a ring modulator effect.

"Bitrate" gives the effect of sample rate reduction.

"Lowres" combines sample rate reduction with bit depth reduction.

"Sine Fold" is a sinewave shaping distortion.

"Scream" produces a less bright type of distortion.

Effect Mod

→ **Set the character of the selected effect algorithm.**

Mix

→ **Set the mix between the dry signal and the effect signal.**

The EQ section



This is a Lo Cut and Hi Cut filter, which lets you cut out bass (lo cut) and treble (hi cut) frequencies from the sound.

Lo Cut

→ **Turn up the Lo Cut knob to cut out bass frequencies from the signal.**
Range: 20.0 Hz to 4 kHz.

Hi Cut

→ **Turn down the Hi Cut knob to cut out treble frequencies from the signal.**
Range: 200 Hz to 20 kHz.

The Send section



The Send knobs can be used for tapping the output signal of the Slot to the corresponding FX Send Out jacks on the rear panel (see "FX Send Out"). You could then route the signals to external effect devices and then further to a separate mixer/audio channel.

→ **Control the Send output levels with the corresponding Send knobs.**

Connections



! **Remember that CV connections are NOT stored in the Mimic patches! If you want to store CV connections between devices, put them in a Combinator device and save the Combi patch.**

Sequencer Control inputs

The Sequencer Control CV and Gate inputs allow you to play Mimic from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity. There are also inputs for modulating the Pitch Bend and Mod Wheel parameters.

CV In 1-4

These four assignable control voltage (CV) inputs can be used for modulating parameters on the Mimic front panel, by selecting “CV In” in the Mod drop-down selector for the desired parameter.

Slot Out 1-8

These outputs are breakout jacks for each individual Slot. Connect these to have the audio from the desired Slot(s) routed to its own stereo output pair.

! **Connecting any of the Slot Outs will automatically remove the corresponding Slot from the Master Audio Outs.**

FX Send Out

Here you can route the audio from all eight Slots to external effect devices for further processing. Since there are no FX Return jacks on Mimic, route the processed signal to a separate mixer/audio channel. You can control the Send level of each Slot with the respective Send knobs (see “The Send section”).

Audio Output

These are the main audio outputs. When you create a new Mimic device, these outputs are auto-routed to the first available Mix Channel in the main mixer. If there is no Mix Channel available, a new one will be automatically created.

Tips and Tricks

Optimizing performance/DSP Load

Under some circumstances Mimic could be quite DSP demanding. If you experience any excessive DSP Load or playback problems, please check the following:

- **Try another Stretch Type (see "Stretch").**
 Especially the "Advanced" Stretch Type can be quite DSP consuming.
- **If you are playing high-pitched notes using a low-pitched sample, the stretch function has to calculate a lot. To reduce the workload, try loading a higher pitched sample (which has a higher Root Key) and use that instead.**
- **Try to avoid having a lot of notes decaying at the same time. Try shortening the Release time of the Amp Envelope (see "The Filter Envelope and Amp Envelope sections") and/or deactivating Play Thru in Slice Mode (see "Play Thru").**

Creating a "velocity layered" instrument

A neat way to simulate a velocity layered instrument is to sample the same note repeatedly, of let's say a bass or piano, and then modulate which of the recorded notes should play back. This will give you a more "live" and organic sound when you play. In this example we have sampled the same note of a bass guitar eight times in a single recording, played with a little different nuances between the eight picks:



1. **Make sure Pitch Mode is selected.**
2. **Click the "Snap to Slices" LED in the Start Position section.**
 Do this to ensure that the playback will start from a slice marker (and not from in between slice markers).
3. **Select "Velocity" in the Start Position Mod drop-down list and turn up the Start Position Mod amount.**
 We are now controlling which "slice" should play back from Keyboard Velocity. You can of course choose another modulation source (e.g. Mod Wheel) if you want to control the start position using the Mod Wheel instead.

- **Make sure you set the Amp Envelope "Release" time so that the consecutive "slice" doesn't play back.**

Extending the sample “tail” (without looping)

Sometimes it could be useful to extend the tail of a sample (a drum sample, for example), to have it decay longer than the original sample length. An easy way of doing this is by modulating the “Speed” parameter as follows:



1. **Turn down the “Speed” knob close to minimum.**
2. **Turn up the “Speed Mod” knob and select “Filt Env” as Speed Mod source.**
3. **Select the “Advanced” stretch type in the Stretch section.**
4. **Raise the Filter Envelope (D)ecay slider to the desired value.**
 When you play and hold a key down, the Speed will quickly raise and then slow down in the sample tail, making the playback time extend.

Automating the sample Start and End markers

Besides the regular modulation capabilities in Mimic, you could also automate and modulate the sample Start and End markers from the main sequencer, to have them reposition in real-time during sequencer playback. This could be useful if you have a longer recording/sample that you want to alter the playback position in:

1. **Record some notes on the Mimic sequencer track in the sequencer and then hit Stop twice.**
2. **Hit Record again in the sequencer and drag the Sample Start and Sample End markers during recording:**



- **If you like, drag and move the Start/End marker up into the Slice Marker area above the waveform, to have the Start/End marker snap to the closest slices. This works in all modes, not just in Slice Mode.**

3. **Hit Stop in the sequencer twice when you are done recording.**
 If you moved both the Sample Start marker and the Sample End marker you will now have two Parameter Automation lanes with clips on them on the Mimic sequencer track:



- **The clips on the Start Pos and End Pos lanes represent the movements of the Sample Start and Sample End markers respectively.**

Chapter 16
Thor Polysonic Synthesizer

Introduction



Thor is an advanced synthesizer with many unique features.

The design could be described as semi-modular, in that the oscillator and filter sections are open slots that allow the user to select between various different oscillator and filter types, each with a distinct character. Some of these designs were inspired by selected vintage equipment.

As a result, Thor is capable of producing an astounding array of sounds.

While it offers a lot of scope for serious sound modeling, it still has a basically simple and user-friendly interface.

In the extensive Modulation bus routing section both audio and control signals (CV) co-exist, and more or less any routing combination can be assigned. Use audio to modulate a CV signal or vice versa - Thor's modulation capabilities are virtually limitless.

Thor also features an advanced step sequencer which can be used for creating melody lines or purely as a modulation source.

There are also audio inputs on the back panel. By connecting the output of another device to these inputs, you can use Thor's filters, envelopes etc. to process the sound, or you can use the external audio source to modulate a Thor parameter.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

About basic synthesizer terminology

This chapter assumes familiarity with common synth terminology like oscillators, waveforms, filters and envelopes. If you are new to Reason (or these terms), you may want to read the "Subtractor Synthesizer" chapter first, where these elements and how they interact are described from a more basic point of view.

Loading and Saving Patches

Loading and saving patches is done in the same way as with any other Reason device - see "Loading patches" and "Saving patches".

Thor elements

In the picture below an unfolded Thor device is shown.



Thor's user interface consists of the following elements (from the top down):

- **The Controller panel, which is always shown if Thor is unfolded.**
 See "The Controller panel".
- **The main Programmer panel contains all the synth parameters.**
 The Programmer can be shown/hidden by clicking the "Show Programmer" button on the Controller panel. See "Using the Programmer".
- **The Modulation bus routing section.**
 See "Modulation bus routing section".
- **The Step Sequencer section, where you can program up to 16 steps to produce short melody lines/grooves or use it as a modulation source.**
 See "Step Sequencer".

The Controller panel



The Controller panel contains standard Master Volume and Pitch and Mod controls, Keyboard Mode/Note Triggering sections and four virtual (freely assignable) controls. The panel also has a patch display and standard Select/Browse/Save patch buttons (these are always shown even if Thor is folded).

The Keyboard Mode section

In this section you make basic keyboard related settings for a patch. It has the following options:

Function	Description
Polyphony	This setting determines the number of voices that you can play simultaneously when Polyphonic mode is selected. The maximum number of voices is 32.
Release Polyphony	This governs the number of voices that are allowed to naturally decay/ring out (in the release phase of the envelope) when new notes are triggered and Polyphonic mode is selected. E.g. if you set this to "0", any new note(s) will cut off the release of any previously triggered notes.
Mono Legato	Mono Legato mode is monophonic regardless of the Polyphony setting. It works as follows: ➔ **Hold down a key and then press another key without releasing the previous.** Notice that the pitch changes, but the envelopes do not start over. That is, there will be no new "attack".
Mono Retrig	Mono Retrig is also monophonic and this mode means that when you press a key the envelopes are always retriggered.
Polyphonic	This is the standard polyphonic play mode - you can play the number of voices set with the Polyphony parameter.
Portamento On/Off/Auto	The knob is used for controlling portamento - a parameter that makes the pitch glide between the notes you play, rather than changing the pitch instantly as soon as you hit a key on your keyboard. By turning this knob you set how long it should take for the pitch to glide from one note to the next as you play them. There are three basic portamento modes: • **In Auto mode, there will only be any portamento when playing more than one note. If any of the Mono modes is selected, portamento will only affect the legato notes.** • **When set to On, portamento is applied to all notes.** • **Off means no portamento.**

Note Triggering section

Using the buttons in this section you can select in what way Thor will respond:

- **Via note input only.**
- **Via the Step Sequencer only (see "Step Sequencer").**
- **Or both.**

The section also has a standard Note On indicator.

About the assignable controls

- **The rotary knobs and buttons in the Controller panel are assignable controls that can be assigned to multiple parameters and functions in Thor.**
- **You assign parameters to the knobs and buttons in the Modulation Routing panel (these are located on the "Modifiers" sub-menu - see "Modulation bus routing section").**
- **Movements of the assignable controls can be recorded as automation.**
- **Each control can be assigned to any number of parameters.**
- **Clicking on the label for a Rotary or Button lets you type in an appropriate name for it.**

About the button key note function

To the right of the two assignable buttons there are corresponding spin controls and displays. These can be used to assign a key for turning the button on momentarily, as long as the key is held down.

→ **Use the spin controls (or click in the display and move the mouse up or down) to assign a key for the button status.**
The assigned key will now turn the function(s) assigned to the button on for as long the key is held down.

→ **Note that the key note function can only switch from off to on, not the other way around, so make sure the button is deactivated if you wish to use this function.**
An assigned key will not trigger a note, only the button status. Also note that the button will not light up when you press the assigned key.

The Pitch Bend and Modulation wheels

- **The Pitch and Mod wheels on the Controller panel will mirror the corresponding actions on your master keyboard.**
- **The Range parameter (like for all instrument devices) sets the range of the Pitch Bend action.**
- **Pitch Bend is pre-wired to the pitch parameter of the three oscillators, but you can of course use it to control any parameter you like. If you don't want Pitch Bend to affect oscillator pitch, simply set the Range parameter to "0".**

Master volume

This is the main volume control for outputs 1 & 2.



Using the Programmer

The Programmer contains the main synth parameters.

- **To show the Programmer panel, click the "Show Programmer" button on the Controller panel.**
 The Programmer appears below the Controller panel.





- **The Programmer panel is divided into two sections; the Voice section to the left and the Global section to the right. The Global section has a separate brown panel to differentiate it from the Voice section.**
 The Voice section contains the basic synth parameters and the parameters are "per-voice", i.e. all envelope and LFO cycles are triggered individually for each voice. The Global section to the right contains global parameters that affect all voices.
- **There are three open Oscillator slots, a Mixer, two open Filter slots, a Shaper, three Envelope generators, an LFO and an Amplifier in the Voice section.**
 The open Oscillator and Filter slots allow you to select between different types of oscillators and filters.
- **The Global section contains a second LFO, a Global Envelope, a third open Filter slot and Chorus and Delay effects.**

Basic connections - a tutorial

There are certain pre-defined connections available between the Oscillator 1-3 slots and the Mixer, Filter 1/Shaper, Filter 2 and Amp sections. On the panel itself, lines with arrows are shown to indicate the standard signal paths.

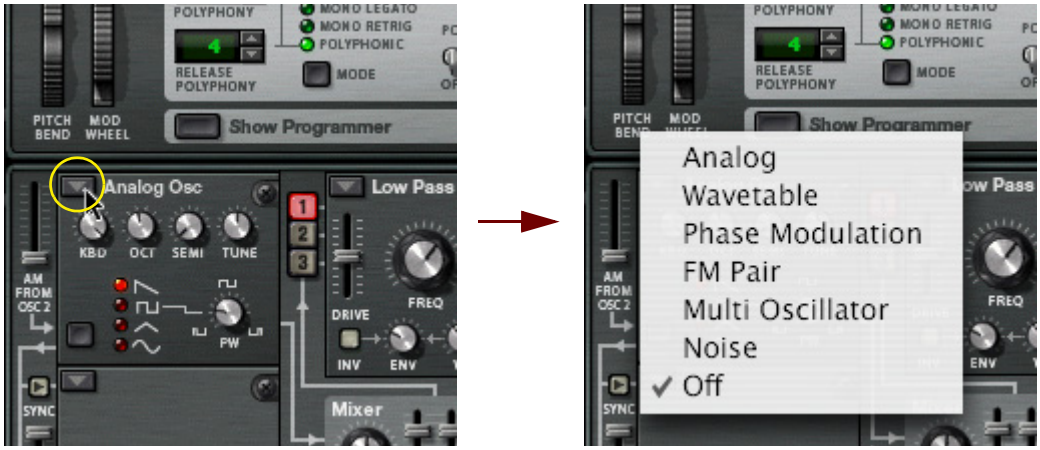
- **Note that you can also connect sections using the Modulation bus section (see "Modulation bus routing section"). You are not in any way limited to the pre-defined routings, but they do provide a quick and convenient way to connect the basic synth "building blocks" together.**

In the following tutorial we will create a standard setup using two oscillators and two filters to demonstrate Thor basics and the (standard) signal path:

1. **Select "Reset Device" from the context menu or from the Edit menu.**
 The "Init patch" is a basic setup with an Analog oscillator in Oscillator slot 1 and a Ladder LP filter in Filter slot 1 loaded. A connection between Oscillator 1, Filter 1 and the Amp section is already activated, so you get a sound when you play.

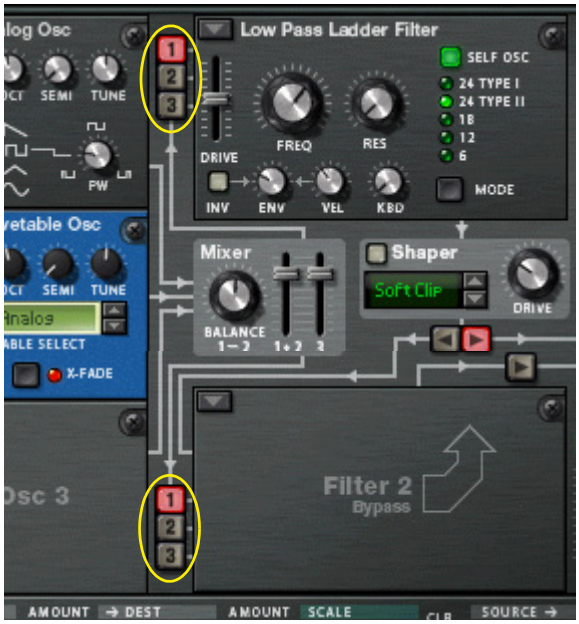
Below the Oscillator 1 slot in the upper left corner are two more slots, currently empty. These are the Oscillator 2 and 3 slots, respectively. The three Oscillator slots are basically identical in that they can each be loaded with one of 6 oscillator types.

2. **Click the arrow pop-up in the upper left corner of the Oscillator 2 slot, and select a second oscillator from the pop-up that appears.**
 The following oscillator types are available; Analog, Wavetable, Phase Modulation, FM Pair, Multi and Noise. For a description of the various oscillator types, see "The Oscillator section".



Selecting oscillator type.

With a basic connection setup, the Oscillator outputs are internally connected to the “Mix” section. To pass the output signal onwards in the signal chain, you first have to activate a connection. This is done using the two vertical rows of routing buttons labelled 1, 2 and 3 to the right of the Oscillator section.



- **The upper row of routing buttons determine which of the Oscillators 1 to 3 are routed to Filter 1, and the lower row which of the Oscillators 1 to 3 are routed to Filter 2.**
 All three oscillators can be simultaneously routed to both filters, serially or in parallel (or any combination of these variations). This is explained later in this tutorial.

By activating one or more of these buttons means that the oscillator (1 to 3) is routed to the corresponding Filter. Currently, Oscillator 1 is connected to Filter 1 slot (which is pre-loaded with a Ladder LP filter).

This is indicated by the “1” routing button being lit. The Filter 2 slot is currently not active, which is indicated by a blank panel.

3. **Click the “2” button to the left of the Filter 1 section so that it lights up to activate a connection for Oscillator 2.**
 Now if you play a few notes you should hear both Oscillator 1 and Oscillator 2, via the Filter 1 section.



- **The Filter 1 output passes via the Shaper (currently not activated), on to the Amp section, and finally to the Main Outputs.**
 Actually, the Amp section output is routed via the Global section before being sent to the Main Outputs, but as currently nothing is activated in the Global section the signal passes through unprocessed.

4. **Next, click the arrow pop-up in the upper left corner of the Filter 2 slot.**
 A pop-up menu with the four available Filter types appears. For a description of the filter types, see "Filter slots".



5. **Select a type of filter, e.g. a Comb filter for the Filter 2 slot.**
 Now that the Filter 2 slot in the Voice section is active, you can connect the oscillators to it by using the lower row of routing buttons.
6. **Click the routing buttons "1" and "2" to the left of the Filter 2 slot so that the buttons are lit.**
 Now the two oscillators are connected to Filter 2.
7. **Make sure the arrow routing button that points to the Amp section just above the Filter 2 section is activated.**
 Now if you play a few notes, both oscillators are routed via both filter sections in parallel. You could of course select to pass only one of the oscillators via one filter and both oscillators via the other - any combination is possible.



You can also connect the Filter 1 and 2 sections serially, meaning that the output of Filter 1 is passed through Filter 2 before reaching the Amp section. This is done as follows:

8. **Switch off the routing buttons "1" and "2" to the left of the Filter 2 slot.**
 If you leave them on the oscillators will pass through Filter 2 twice; both via Filter 1 and directly. This is also perfectly "allowable", but to make things clearer in this tutorial we will use a standard serial filter setup.
9. **Click the Arrow "left" button below the Shaper.**
 Now the filters are connected serially, with the output of Filter 1 (via the for now inactive Shaper) being connected to the Filter 2 input. Both oscillators are processed by both filters connected in series.



That concludes this tutorial on how the pre-wired connections in the Voice section can be used, but note that you can also use the Modulation bus to make connections - see "Modulation bus routing section".

Other pre-defined routing assignments

There are other sections in Thor which are pre-defined and can be used without having to make any prior assignments:

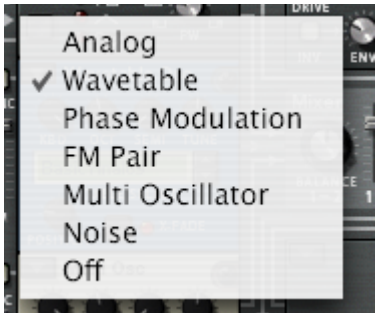
- **The Amp Envelope and the Filter Envelope control the volume level and frequency of the Filters (1 & 2), respectively.**
 The amount of filter envelope control is controllable by using the “Env” parameter in each Filter section.
- **The effects (Delay/Chorus) in the Global section are part of the signal chain and can simply be switched on and used.**

The Oscillator section



Oscillators generate the basic raw sound (pitch and waveform) that can in turn be processed by the other parameters. The Oscillator section contains three open slots which can each be loaded with one of six oscillator types. The three Oscillator slots are numbered 1-3, with the top slot housing Oscillator 1, the middle slot Oscillator 2 and the bottom slot Oscillator 3.

- **The Arrow button in the top left corner of each slot opens a pop-up menu where an oscillator type can be selected for the corresponding slot.**



There are six Oscillator types available:

- **Analog**
- **Wavetable**

- **Phase Modulation**
- **FM Pair**
- **Multi Oscillator**
- **Noise**

You can also select Off mode (no oscillator).

Common parameters

The specific parameters of the various oscillator types are described separately, but there are also common parameters that apply to all oscillator types. These are:

- **Octave (OCT) knob - this changes the pitch of the oscillator in octave steps.**
 The range is ten octaves.
- **The Semi knob changes the pitch of the oscillator in semi-tone steps.**
 The range is 12 semitone steps (1 octave).
- **The Tune knob fine tunes the pitch of the oscillator in cent steps.**
 The range is +/- 50 cents (down or up half a semitone).
- **Keyboard Track (KBD) - this knob sets how much the oscillator pitch tracks incoming note data.**
 Turned fully clockwise the pitch tracks the keyboard normally, i.e. a semitone per key.
- **All oscillators also have waveform selectors and a modifier parameter. How the waveform selection works, and what parameter is the modifier varies according to the selected oscillator type.**
- **Important to note is that if you have made a modulation routing to an oscillator parameter e.g. the modifier, and then change the oscillator type, the modulation will be transferred to the corresponding parameter in the new oscillator.**
 The same goes for all common parameters (tuning and tracking). If you switch oscillator type, all common parameters are left unchanged.
- **Oscillators can be synced - see "About Oscillator Sync".**
- **Any oscillator type loaded into the Oscillator 1 slot can also be amplitude modulated by Oscillator 2 - see "About Amplitude Modulation (AM)".**

Analog oscillator



This is a classic analog oscillator with 4 standard waveforms. The waveform selector button is in the lower left corner of the oscillator panel, but you can also click directly on the waveform symbols to switch waveform. The four available waveforms are from the top down (as displayed on the panel): Sawtooth, Pulse, Triangle and Sine.

- **The Mod parameter (PW) controls pulse width and only affects the pulse waveform.**
 By modulating the PW parameter the width of the pulse wave changes, allowing for PWM (Pulse Width Modulation) which is a standard feature in most vintage analog synths.

→ **For a perfect square wave, set pulse width (PW) to 64.**

Wavetable oscillator



Wavetable oscillators has been the basis of several vintage synths (PPG, Korg Wavestation and many others).

- **With the Wavetable oscillator, you select between 32 wavetables, where each wavetable contains several (up to 64) different waveforms. By using an envelope or a LFO you can sweep through a wavetable to produce timbre variations.**

The parameters are as follows:

- **Position is the modifier (Mod) parameter and controls the position within the selected wavetable, i.e. which waveform is active at a given time.**
 By modulating the Position you can sweep through the waveforms in the selected wavetable. You can of course also use a single static waveform in a wavetable if you so wish, by not applying any modulation to this parameter.
- **The X-Fade button determines whether the change between waveforms in a wavetable should be abrupt (X-Fade off), or smooth (X-Fade on).**
 If set to on, the waveform transitions are cross-faded.
- **There are 32 wavetables that can be selected using the up/down buttons or by clicking in the Wavetable display.**

Some of the wavetables have waveforms that sequentially follow the harmonic series, i.e. each following waveform adds a harmonic. Others have waveform series that produce a sound similar to oscillator sync when swept, and other wavetables are simply mixed waveforms. The last 11 wavetables are based on wavetables used in the original PPG 2.3 synthesizer.

Phase Modulation oscillator



The Phase Modulation oscillator is inspired by the Casio CZ series of synthesizers. Phase modulation is based on modulating the phase of digital waveforms to emulate common filter characteristics.

- **You have a First and Second waveform which can be combined. Instead of mixing the two waveforms they are played in series, one after the other.**
 This adds a fundamental one octave below the pitch of the original sound.
- **The PD parameter (Mod) changes the shape of the wave, much like a filter does.**

The following waveforms (sequentially from the first) are available as the First waveform:

- **Sawtooth**
- **Square**

- **Pulse**
- **Pulse and Sine**
- **Sine and flat (half sine)**
- **Saw x Sine**
- **Sine x Sine**
- **Sine x Pulse**

The last three waveforms could be described “resonant”, as these originally were meant to simulate filter resonance. They didn’t really do this very accurately, but nevertheless constituted an important part of the sound.

The Second waveform has the same available waveforms except the last three, and it can also be bypassed altogether. You can combine waveforms freely, except it is not possible to combine two “resonant” waveforms.

FM Pair oscillator



As the name implies, this oscillator generates FM, where one oscillator (Carrier) is frequency modulated by a second oscillator (Modulator). Although very simple to use (unlike most hardware FM synths), this oscillator can produce a very wide range of FM sounds.

- **The Carrier and Modulator selector buttons set the frequency ratio between these two oscillators (the range is 1-32).**
 The frequency ratio is what determines the basic frequency content, and thus, the timbre of the sound.
- **The FM knob sets the amount of frequency modulation.**
 This is also the Modifier parameter. If FM amount is set to zero, there is no FM and the output will be a pure sine wave.

- **If you set FM Amount to zero and step through the values of the Carrier oscillator, you can hear that the pitch is changed according to the harmonic series.**
- **Stepping through the Mod oscillator values will change the pitch in the same way, although FM Amount has to be set to a value other than zero to be able to hear it.**

Thus, 2:2 is the same wave shape as 1:1 but one octave higher in pitch, 3:3 is the same wave shape as 2:2 but a fifth higher in pitch and so on.

Multi oscillator



This versatile oscillator can simultaneously generate multiple detuned waveforms (of a set type) per voice. It is great for producing complex timbres e.g. to simulate cymbal or bell sounds, but can also generate a wide range of harmonic sounds.

- **The following basic waveforms are available: Sawtooth, Square, Soft Sawtooth, Soft Square, Pulse.**
 You switch waveforms using the button in the lower left corner, or by clicking directly on the waveform symbol.
- **The Amount (AMT) parameter governs the amount of detune.**
 Turn clockwise for more detune. This is also the modifier (Mod) parameter. Using low Amount settings can produce subtle detune variations that makes the sound shift and move endlessly, like an advanced chorus effect, whereas higher Amount settings can produce wild, detuned timbres.
- **The Detune Mode parameter sets the basic operational mode of the detuning.**
 If Amount is set to 0, only the "Octave" and "Fifth" Detune modes actually change the sound, as these modes start off with dual waveforms tuned one octave and a fifth apart, respectively. The "Fifth Up" and "Oct UpDn" modes detune waveforms as the names imply between zero to full Amount settings. "Linear" will change the amount of detune according to where on the keyboard you play; in lower keyboard ranges the amount of detune is stronger than in higher keyboard ranges and vice versa. The other modes (Interval and Random) basically add multiple waveforms and detune them in various ways that will produce different results.

Noise oscillator



The Noise oscillator can not only produce white and colored noise, but can also be used either as a pitched oscillator or as a modulation source.

It has the following basic parameters:

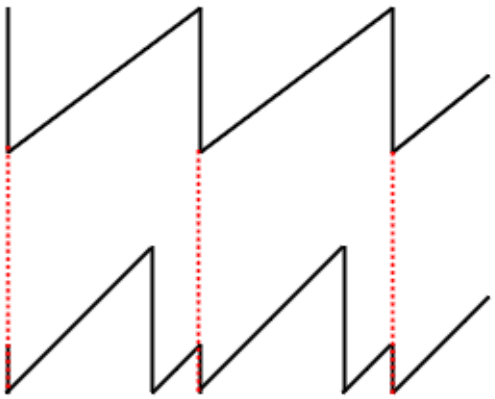
- **There is a single Noise parameter (apart from the standard tuning and kbd track knobs).**
 This is the Noise modifier parameter, that controls different parameters depending on the selected Oscillator mode, see below.

→ **The Waveform selector button in the bottom left corner is used to set the Oscillator mode.** The following modes are available:'

Mode	Description
Band	In this mode, the Oscillator knob controls bandwidth. Turned fully clockwise, the oscillator produces pure noise. Turning the knob counter-clockwise gradually narrows the bandwidth until a pitch is produced. The pitch will track the keyboard normally if the keyboard (KBD) knob is set fully clockwise.
S/H	S/H stands for "sample and hold", which is a type of random generator. The Oscillator knob controls the rate of the sample and hold. With high Oscillator knob settings, it produces colored noise with a slightly "phased" sound quality. With lower rate settings you can use the oscillator as a modulation source like a LFO with random values. For example, if you modulate the pitch of another oscillator using S/H with a low Rate setting as the source, you will get stepped random modulation of the pitch.
Static	As the name implies, this can generate the sound of static interference if you use low Oscillator settings. The Oscillator parameter controls Density, i.e. the amount of static. High Density settings generates noise.
Color	This produces colored noise, which is basically noise where certain frequency areas are filtered, i.e. cutting or boosting certain frequency areas in the noise. The Oscillator knob controls Color. With a maximum Color setting you get white noise, and lower settings produces noise emphasizing lower frequencies.
White	This produces pure white noise, where all frequencies have equal energy. There is no associated Oscillator parameter for White noise.

About Oscillator Sync

Oscillator sync is when one oscillator will restart the period of another oscillator, so that they will have the same base frequency. If you change or modulate the frequency of the synced oscillator you get the characteristic sound associated with oscillator sync.



Syncing oscillator

Synced oscillator

In Thor, oscillator 1 is always the syncing oscillator, i.e. oscillator 1 controls the base pitch of oscillators 2 and 3, which are the synced oscillators.

- **Switch Oscillator Sync on or off by activating the Sync buttons to the left of Oscillator slots 2 and 3.**



- **The Sync "BW" sliders to the left of Oscillator slots 2 and 3 allows you to adjust the sync bandwidth.**
 This allows you to change the character of the oscillator sync. The parameter basically sets how abrupt the reset is - high bandwidth settings produces a more pronounced sync effect and vice versa. The picture above illustrates high bandwidth reset - if lower bandwidth settings are used the synced osc curve will be more rounded at the reset points.

About Amplitude Modulation (AM)

AM (Amplitude Modulation) is often referred to as ring modulation. AM works by multiplying two signals together.

- **In Thor, Oscillator 2 amplitude modulates Oscillator 1.**
 The Ring Modulated output will then contain added frequencies which are generated by the sum of, and the difference between the two signals. This can be used for creating complex, enharmonic sounds.
- **The amount of AM is set using the slider to the left of the Oscillator 1 slot.**



Mix section



The Mix section allows you to adjust the levels and the relative balance of the three oscillators.

- **The two sliders controls the output levels of oscillators 1-2 and oscillator 3, respectively.**
- **The Balance knob sets the balance between oscillator 1 and 2.**
 The Balance parameter is also a modulation destination, allowing you to modulate the balance of the two oscillators with e.g. an LFO. Note that the oscillators have to be connected to the filter(s) via the numbered routing buttons for the Mix section settings to have any effect.

Filter slots



Thor has three open Filter slots, two in the Voice section (which act per-voice) and one in the Global section which is global for all voices (see "Global Filter slot").

- **You select (or change) filter type for a slot by clicking the arrow button in the top left corner of a slot.**
 On the pop-up you can select between 4 filter types and bypass mode. Available filter types are Ladder LP, State Variable, Comb and Formant, each described separately below.

The following general rules apply:

- **Filters are pre-wired to the Filter Envelope (see "Filter Envelope").**

- **Filters 1 & 2 can be used serially (i.e. the output of Filter 1 goes (via the Shaper) to the input of Filter 2, or in parallel (meaning that one signal goes to Filter 1 and another to Filter 2).**

The pre-defined routings of the three oscillators into the Filter 1-2 sections is described in the "Basic connections - a tutorial" passage.

Common parameters

As with the open oscillator slots, there are certain parameters which are common for all filter types.

These are as follows:

- **All the filter types have large knobs for the filter frequency (FREQ) parameter and the filter resonance (RES) parameter.**
 This works slightly differently for the Formant filter - see "Formant filter".
- **The "KBD" parameter sets how the filter frequency tracks incoming note pitch data.**
 Some filter types (Ladder/State Variable/Comb) can "self oscillate" and be used as extra oscillator sources.
- **The "ENV" parameter sets how much the filter frequency responds to the Filter Envelope.**
- **The "VEL" parameter sets how much incoming note velocity affects the Filter Envelope Amount.**
 In other words, for this parameter to have any effect it requires that the "ENV" parameter is set to a value other than zero.
- **The "INV" button inverts how the filter frequency responds to Envelope settings.**
- **The "Drive" parameter allows you to adjust the input gain to the filter.**
 By driving the filter harder you can add further character to the sound.
- **Any parameter settings, as well as any modulation assigned to parameters, will be kept even if you change the filter type.**

Ladder LP Filter



The Ladder LP filter is a low-pass filter inspired by the famous voltage controlled filter patented by Dr. Robert Moog in 1965. The name originates from the ladder-like shape of the original transistor/capacitor circuit diagram.

The original filter also had certain non-linear characteristics which contributed to the warm, musical sound it is renowned for. These characteristics are faithfully reproduced in the Ladder LP filter.

There is also a built-in shaper in the feedback (self-oscillation) loop. If self-oscillation is activated (see below), the shaper will distort the sound to produce these non-linear characteristics. To adjust the intensity of this distortion you use the Drive parameter.

- **There are 4 different Filter slopes available; 24, 18, 12 and 6 dB/oct.**
 24dB slope comes in two different types:
- **Type I - The shaper (controlled with the Drive parameter) is placed at the filter output but before the feedback loop.**
- **Type II - The shaper (controlled with the Drive parameter) is placed at the filter input after the feedback loop.**

Note that "Self Osc" (see below) must be activated for the shaper to operate.

- **This filter can self-oscillate and will produce a playable note pitch with high Resonance settings if this is activated.**
 Self-oscillation can be switched on or off by using the "SELF OSC" button. The "KBD" knob governs how the frequency tracks the keyboard, turned fully clockwise will produce 12 semitones/octave tracking.

State Variable Filter



This is a multi-mode filter which offers 12 dB/octave slope Lowpass (LP), Bandpass (BP), Highpass (HP), plus Notch and Peak filter modes which are sweepable between HP/LP states, similar to the vintage Oberheim SEM filter.

The filter modes are as follows:

- **LP 12 (12 dB lowpass)**
 Lowpass filters let low frequencies through and cut off high frequencies. This filter type has a 12dB/Octave slope.
- **BP 12 (12 dB bandpass)**
 Bandpass filters cut both high and low frequencies, leaving the frequency band in between unaffected. Each slope in this filter type is 12 dB/Octave.
- **HP 12 (12 dB highpass)**
 Highpass filters let high frequencies pass and cut off low frequencies. This filter type has a 12dB/Octave slope.
- **The "Notch" and "Peak" filter modes employ a combination of two outputs from the same filter combining LP and HP set to the same the filter frequency.**
 The "LP/HP" knob associated to these two filter modes can modulate the state of the filter from low-pass to high-pass. If the knob is in the mid-position, you get a Peak or Notch filter slope (depending on the mode). The HP/LP parameter can be assigned as a modulation destination.
- **This filter can self-oscillate and will produce a pitch with high Resonance settings if this is activated.**
 Self-oscillation can be switched on or off by using the "SELF OSC" button. The "KBD" knob governs how the frequency tracks the keyboard, turned fully clockwise will produce 12 semitones/octave tracking.

Comb filter



The Comb filter can add subtle pitch variations and phasing-like effects to sounds.

- **Comb filters are basically very short delays with adjustable feedback (controlled with the Resonance knob).**
 A comb filter causes resonating peaks at certain frequencies. Comb filters are used in various signal processing devices like flangers, and produces a characteristic swooshing sound when the frequency is swept.
- **The difference between the "Comb +" and "Comb –" modes is the position of the peaks in the spectrum.**
 The main audible difference is that negative Comb mode causes a bass cut.
- **The Resonance parameter in both cases controls the shape and size of the peaks.**
 This filter will produce a pitch with high Resonance settings combined with low frequency settings.

Formant filter



The Formant filter type can produce vowel sounds. There are no Frequency or Resonance parameters, instead you have a horizontal "X" parameter slider and a vertical "Y" parameter slider that operate together to produce the various filter formant characteristics.

- **You can alter the settings of both the "X" and "Y"parameters simultaneously by moving the "dot" inside the gray rectangle on the filter panel.**
 Horizontal movement changes the "X" parameter, and vertical movement the "Y" parameter.



- **The ENV-VEL-KBD knobs affect the "X" parameter.**
 The parameter can be CV controlled.
- **The "Gender" parameter changes the basic timbre of the vowel generation between male (low Gender settings) and female (high Gender settings) voice characteristics.**
 Gender can also be CV controlled.

Shaper

Waveshaping is a synthesis method for transforming sounds by altering the waveform shape, thereby introducing various types of distortion. The Shaper can radically transform the sound or just add a little warmth, depending on the mode and other settings.

- **The Shaper input is taken from the Filter 1 output.**
- **The Shaper is activated with the button in the top left corner of the section.**



- **The Drive parameter sets the amount of waveshaping.**
 Tip: By raising the Filter 1 Drive parameter you can add even more grit and distortion to the Shaper output.
- **The Shaper has 9 modes, selectable with the spin controls or by clicking in the Mode display, all which distort the waveform in various different ways.**
 These modes are; Soft and Hard clip, Saturate, Sine, Bipulse, Unipulse, Peak, Rectify and Wrap. Exactly how the various modes affect the sound depends on many factors, and there is a slightly random element to the resulting distortion. We recommend simply trying the different modes to hear what happens - many interesting types of distortion of the original signal are guaranteed!

Amp section



The Amp (amplifier) section has two inputs (from Filter 1 & 2) and one output that is routed to the Global section (and on to the Master Level and the Main Outputs).

- **The Gain knob controls the level and the Velocity knob controls the Gain modulation, i.e. how much velocity affects the level - positive values means that you get higher level the faster you strike a key.**
- **The Pan knob controls the relative stereo position of the individual voices.**
 By applying modulation to this parameter, you can make individual voices appear in different stereo positions when you play.

LFO 1



An LFO (Low Frequency Oscillator) is used for generating cyclic modulation. A typical example is to have an LFO modulate the pitch of an oscillator to produce vibrato, but there are countless other applications for LFOs.

- **LFO 1 will apply modulation polyphonically.**
 I.e. if LFO 1 modulation of a parameter is assigned, an individual LFO cycle will be triggered for each note you play.
- **You select a LFO waveform by using the spin controls beside the waveform display, or by clicking in the display and moving the mouse up or down.**

The following parameters are available for LFO 1:

Parameter	Description
Rate	This sets the frequency or rate of the LFO.
Waveform	This sets the LFO waveform. Apart from standard waveforms (sine, square etc.) there are various different random, non-linear and stepped waveforms. The shape of the waveforms are shown in the display, and these shapes basically reflect how a signal is affected.
Delay	This introduces a delay before the LFO modulation onset after a note is played. Turn clockwise for longer delay.
KBD Follow	This determines if (or how much) the Rate parameter is affected by note pitch. If you turn the knob clockwise, the modulation rate will increase the higher up on the keyboard you play.
Key Sync	As explained previously, LFO 1 is polyphonic and will produce a separate LFO cycle for each note played. If Key Sync is off, the cycles are free running, meaning that when you play a note the modulation may start anywhere in the LFO waveform cycle. If Key Sync is on, the LFO cycles are reset for each note played.
Tempo Sync	If this is on, the Rate will be synced to the sequencer tempo.

Envelope sections



There are three Envelope generators in the Voice section. These are the Amp envelope, the Filter envelope and the Mod envelope. Each voice played has a separate envelope. There is also an additional Global Envelope which is described separately - see "Global Envelope".

- **The Filter envelope is pre-wired to control the frequency of Filter 1 and 2.**
 Note that envelope control of filter frequency can be switched off in each Filter section (the Env parameter can be set to 0), so the Filter Envelope can be used to control other parameters as well.
- **The Amp Envelope is pre-wired to control the amplitude (volume).**
 Similarly, the Amp envelope can also be used to control other parameters, but in the Voice section you cannot switch off or bypass the Amp Envelope - if no voice is active (i.e. if there is no gate trigger input to the Amp envelope) there will be no output from oscillators or any external audio source routed to the Voice section.
- **The Mod Envelope can be freely assigned to control parameters.**
 This is done in the Modulation section.

Filter Envelope

The Filter Envelope is a standard ADSR envelope as used in the Subtractor.

- **By setting up a filter envelope you control the how the filter frequency or some other parameter should change over time with the four parameters, Attack, Decay, Sustain and Release.**
 Please refer to the Subtractor chapter for a description of these parameters.
- **The "Gate Trig" button can be used to switch off the envelope triggering from notes (which is the normal mode) and allow the envelope to be triggered by some other parameter.**
 "Gate Trig" should normally be activated.
- **The time ranges of each step are as follows:**
 Attack: 0 ms - 10,3 s / Decay and Release: 3 ms - 29,6 s. Sustain is not set as a time but as a level (from Off to 0dB).

Amp Envelope

The Amp Envelope is also a standard ADSR envelope.

- **By setting up a Amp envelope you control the how the amplitude or some other parameter should change over time with the four parameters, Attack, Decay, Sustain and Release.**
 Please refer to the Subtractor chapter for a description of these parameters.
- **The "Gate Trig" button can be used to switch off the envelope triggering from note input (which is the normal mode) and allow the envelope to be triggered by some other parameter.**
 "Gate Trig" should normally be activated.
- **The ranges of each step are the same as for the Filter envelope.**

Mod Envelope

This is a general purpose ADR (Attack, Decay, Release) envelope with a pre-delay stage before the Attack phase. The Delay to Decay phase can also be looped. Apart from standard Attack, Decay and Release stages the Mod Env has the following parameters:

Parameter	Description
Delay	This can set a delay before the onset of the envelope.
Loop	If this is activated, the envelope phase from Delay to Decay will continuously loop.
Tempo Sync	If this is on, each stage will have a length that corresponds to beat increments of the current sequencer tempo. E.g. you can have a 1/4 delay before a 1/16 attack phase followed by a 1/8 decay. Each stage can be set a range from 1/32 to 4/1 (4 bars). If this is off, the envelope times are free running and can be set in seconds (same time ranges as for the Filter Envelope).
Gate Trigger	The "Gate Trig" button can be used to switch off the envelope triggering from notes (which is the normal mode) and allow the envelope to be triggered by some other parameter. "Gate Trig" should normally be activated.

Global section



The Global section contains parameters that affect all voices. It contains two effects, an open filter slot, the Global Envelope and LFO 2.

Effects section

There are two global mono in/stereo out effects, a Delay and a Chorus. These effects affect all voices coming from the Amp section equally if activated. The effects are placed after the Global Filter in the signal chain.

- **There are controls for standard Delay/Time and Feedback parameters.**
 Chorus vs. Delay differ only in the delay time range - Chorus is for chorus effects, i.e. short delays, whereas Delay produces echo effects.
- **Delay Time can be Tempo Synced.**
 This is set with the Tempo Sync button - if on the delay time is set in beat resolutions synced to the main sequencer tempo.
- **The Delay and Chorus effects can also be pitch modulated by a built in LFO (the "Mod" parameters).**
 "Rate" controls LFO speed and "Amount" the Stereo width.
- **Dry/Wet governs the balance between the unprocessed (dry) signal and the effect (wet) signal.**

Global Filter slot

This is the Filter 3 slot which can be loaded with one of the filter types. Filter 3 is basically set up as the other filter slots. The difference is that all voices are mixed together before entering the filter. The "ENV" parameter governs modulation by the Global Envelope. If you play one note the filter envelope will trigger. Adding new notes while a note is still held down (legato) will not trigger the filter envelope.

See "Filter slots" for a description of the filter types.

Global Envelope

The Global Envelope 4 is an advanced envelope that is free to use for whatever purpose, but remember it is "single trigger" so it will not retrigger legato notes as explained above. It is an ADSR envelope with a pre-delay stage and a hold stage before the decay phase. You can make it Loop and Sync the time settings to the song tempo.

Apart from standard ADSR parameters, the Global Envelope has the following parameters:

Parameter	Description
Delay	This can set a delay before the onset of the envelope.
Loop	If this is activated, the envelope phase from Delay to Decay will continuously loop.
Hold	This allows you to set a "hold" phase before the Decay.
Tempo Sync	If this is on, each stage will have a length that corresponds to beat increments of the current sequencer tempo. E.g. you can have a 1/4 delay before a 1/16 attack phase followed by a 1/8 decay. Each stage can be set a range from 1/32 to 4/1 (4 bars). If this is off, the envelope times are free running and can be set in seconds (same time ranges as for the Filter Envelope).
Gate Trigger	The "Gate Trig" button can be used to switch off the envelope triggering from notes and allow the envelope to be triggered by some other parameter. This button is normally activated.

LFO 2

→ **The LFO 2 is a standard LFO but is not polyphonic like LFO 1. It is not assigned to any parameter in an "Init" patch so you have to use the Modulation Routing section to use it.**

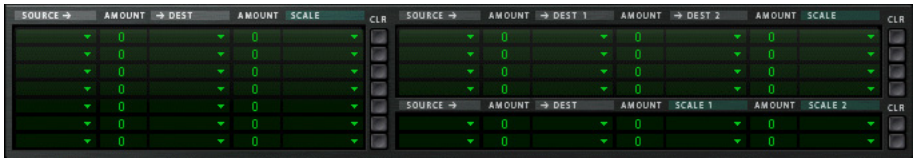
- **Also the LFO 2 "Delay" and "Key Sync" parameters are single trigger, i.e. the LFO will not retrigger these parameters for legato notes.**

→ **You select a LFO waveform by using the spin controls beside the waveform display, or by clicking in the display and moving the mouse up or down.**

The following parameters are available for LFO 2:

Parameter	Description
Rate	This sets the frequency or rate of the LFO.
Waveform	This sets the LFO waveform. Apart from standard waveforms (sine, square etc.) there are various different random, non-linear and stepped waveforms. The basic shape of the waveforms are shown in the display, and illustrate how a signal is affected.
Delay	This introduces a delay before the LFO modulation onset after a note is played. Turn clockwise for longer delay.
Key Sync	If Key Sync is off, the LFO cycle is free running, meaning that when you play a note the modulation may start anywhere in the LFO waveform cycle. If Key Sync is on, the LFO cycle is reset for each note played.
Tempo Sync	If this is on, the Rate will be synced to the sequencer tempo in beat increments (4/1 to 1/32).

Modulation bus routing section



A modulation bus is used to connect a modulation source to a modulation destination. Both audio signals and control (CV) parameters are available. This creates a flexible routing system that complements the pre-wired routing in the Voice panel.

Basic operation - simple tutorial

To illustrate the basic operation of the modulation bus section, let's set up a simple source to destination modulation assignment:

! **If you currently have unsaved settings you wish too keep, don't forget to save them first.**

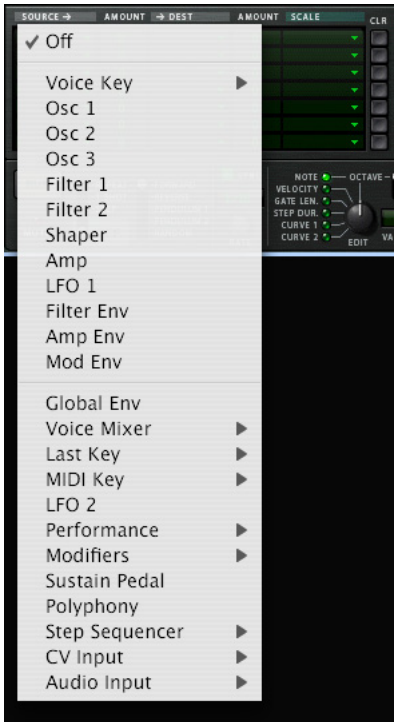
1. **Select "Reset Device" from the Edit menu.**

The Init patch is a simple 1 oscillator/1 filter setup, which produces sound when you play, and will serve the purpose of this tutorial.

- **The left half of the modulation section contains 5 columns, Source, Amount, Dest, Amount and Scale.** Below the column headers there are 7 rows. Each row is a modulation bus where you can have a Source to Destination modulation assigned.

2. **Click in the top row of the leftmost Source column.**
 A pop-up menu appears listing all available Source modulation parameters.

The upper half of the menu contains Voice section source parameters, and the lower half contains various global play and performance-oriented source parameters and the Global Envelope, as well as the Step Sequencer, CV and Audio inputs.



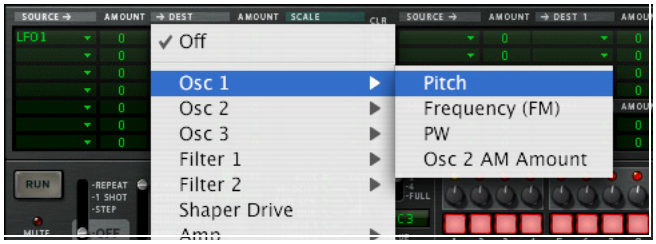
3. Select “LFO 1” from the pop-up.

This means that LFO 1 is the modulation Source, and this can now be assigned to modulate a Destination parameter.

4. Pull down the “Dest” column pop-up in the top row.

A pop-up menu appears listing all available modulation Destinations. The upper half of the menu contains Voice section destinations, and the lower half contains Global section destinations, as well as the Step Sequencer, CV and Audio outputs.

5. Select “Osc 1” from the menu and then “Pitch” from the submenu.



This means that Osc 1 pitch is now assigned to be modulated by LFO 1. Next step is to set the amount of modulation to be applied.

6. Click in the top row Amount column to the right of the Source column, and move the mouse pointer up and down to set an Amount value.
Both positive and negative Amount values can be set (+/- 100%).

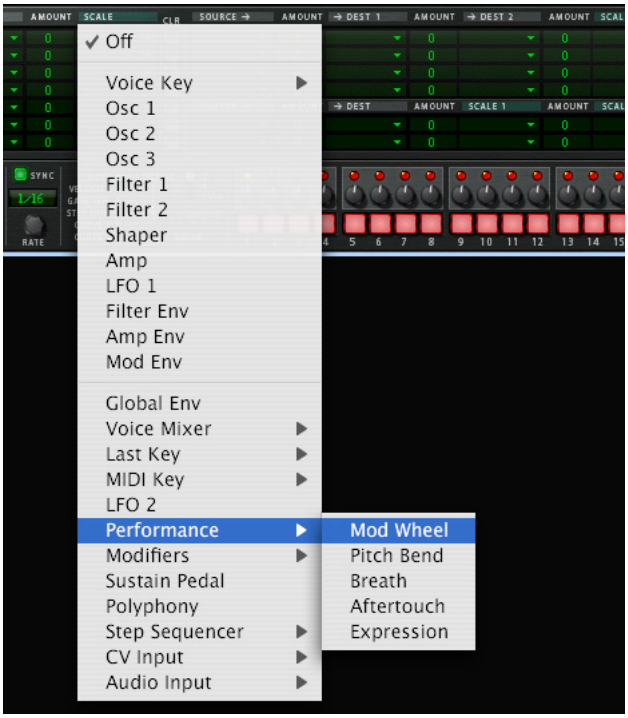
→ **If you now play a few notes you can hear the oscillator pitch being modulated by the LFO to produce vibrato.**
But the vibrato will be constant, which you probably don’t want. This is solved by assigning a Scale parameter, which allows you to assign another parameter to control the modulation Amount.

7. **Pull down the "Scale" column pop-up in the top row.**

A pop-up menu appears listing all available Scale parameters. The upper half of the menu contains Voice section parameters, and the lower half contains various play and performance-oriented parameters and the Global Envelope, as well as the Step Sequencer, CV and Audio inputs.

A typical controller for vibrato is the Mod wheel.

8. **Select "Performance" from the menu and then "Mod wheel" from the submenu.**



This means that Osc 1 pitch is now assigned to be modulated by LFO 1, and the amount of modulation is controlled by the Mod wheel. How much the Scale parameter controls the Amount is set using the "Amount" column for the top row (to the left of the Scale column).

9. **Click in the top row Amount column and move the mouse pointer up and down to set an Amount value.**
 Both positive and negative Scale Amount values can be set (+/- 100%). To fully control the LFO modulation so that there is no vibrato when the Mod wheel is set to zero, set the Amount to 100%.



10. **The modulation routing is now complete!**
 You now have full control over the vibrato modulation by using the Mod wheel.

- **How much modulation will be applied when the Scale parameter is set to maximum is governed by the Source to Destination Amount parameter.**
- **How much the Scale parameter controls the modulation is set with the Scale Amount parameter.**
- **To clear any assigned modulation routing you can use the "CLR" button to the right of the corresponding bus.**

About the three modulation routing types

As described in the tutorial, the principal operators of the Modulation bus routing system are as follows:

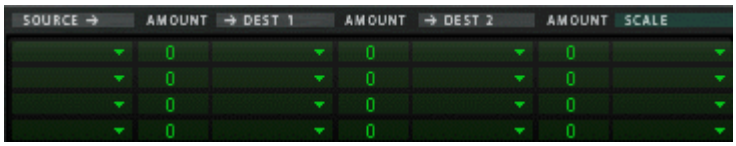
- **You have Modulation Source, Modulation Destination and Modulation Amount parameters.**
- **Optionally, you have a Scale parameter controlling the Modulation Amount, and a Scale Amount that governs how much the Scale parameter controls the Modulation Amount.**

There are three different types of modulation routing busses available in Thor:

- **You have seven "Source –> Destination –> Scale" routing busses.**
 These are the seven rows in the left half of the Modulation section, as covered in the tutorial.



- **There are four "Source –> Destination 1 –> Destination 2 –> Scale" busses.**
 These are the four top rows in the right half of the Modulation section. This works after the same principle but the Source parameter can affect two different Destination parameters (with variable Amount settings) and a Scale parameter that affects the relative modulation Amount for both Destinations.



- **Lastly, there are two "Source –> Destination –> Scale 1 –> Scale 2" busses.**
 This means that a modulation Amount can use two Scale parameters.

 An example: You have the Mod Envelope as Source and Oscillator Pitch as the Destination (Amount set whatever you like). As the first Scale parameter we use the Mod Wheel (Amount set to 100 so that no modulation is applied when the Mod wheel is at zero), and LFO 1 as the second Scale parameter (Amount set to whatever you like). When you move the Mod wheel, the pitch modulation amount will be modulated by both the Mod Envelope and LFO 1 simultaneously.



Modulation Sources - Voice section

The following parameters can be used as Voice section modulation Sources:

Parameter	Description
Voice Key	Voice Key lets you assign modulation according to notes. There are 4 modes selectable from the sub-menus: • **Note - this is keyboard tracking. If a positive Amount value is used and the destination is filter frequency, the filter frequency will track the keyboard, i.e. increase with higher notes.** • **Note2 - this works similarly to Note but within a repeated octave range. E.g. if Note2 modulates Amp Pan the pan position will move from left to right within an octave range then start over. If you play chords normally over the keyboard the effect will be that notes are randomly spread across the stereo field.** • **Velocity - this applies modulation according to velocity (how hard or soft you strike the keys).** • **Gate - this is Gate on/off. E.g. if applied to oscillator pitch you will get one pitch value (set by Amount) when a key is pressed, and another value (the unmodulated pitch) when the key is released.**
Osc 1/2/3	This allows you to route the audio output from the oscillators to a destination.
Filter 1/2	This is the audio output of the filters. All filter parameters affect the destination.
Shaper	This is the audio output of the Shaper module. Note that anything connected to the Shaper, e.g. Filter 1, affects the Shaper output, and thus the resulting modulation.
Amp	This is the audio output of the Amp Gain section.
LFO 1	This allows you to modulate parameters with LFO 1.
Filter Envelope	This allows you to modulate parameters with the Filter Envelope.
Amp Envelope	This allows you to modulate parameters with the Amp Envelope.
Mod Envelope	This allows you to modulate parameters with the Mod Envelope.

Modulation Sources - Global

The following parameters can be used as Global section modulation Sources:

Parameter	Description
Global Envelope	This allows you to modulate parameters using the Global Envelope.
Voice Mixer	This allows you to modulate parameters using the Left and Right Mixer inputs.
Last Key	This will apply modulation according to the last note played (monophonic), either via MIDI, or from the Step Sequencer. For example, you can use Last Key to make a filter's frequency track notes played by the Step Sequencer.
MIDI Key	This applies modulation according to notes globally, not per-voice so in other words it is monophonic. E.g. if you use MIDI Note as Source and a self-oscillating filter's frequency as the destination, the filter will track but you will only be able to play one voice at a time. MIDI Note is handy for transposing Step patterns in real time. There are 3 modes selectable from the sub-menus: • **Note - this is keyboard tracking. If a positive Amount value is used and the destination is filter frequency, the filter frequency will track the keyboard, i.e. increase with higher notes.** • **Velocity - this applies modulation according to velocity (how hard or soft you strike the keys).** • **Gate - this is Gate on/off. E.g. if applied to oscillator pitch you will get one pitch value (set by Amount) when a key is pressed, and another value (the unmodulated pitch) when the key is released.**
LFO 2	This allows you to modulate parameters with LFO 2.
Performance parameters	On this sub-menu you can assign the one of the standard Performance controllers to modulate/scale parameters; Mod Wheel/Pitch Bend/Breath/AfterTouch/Expression.
Modifiers	This is where you assign parameters and functions to be controlled with the virtual 2 Rotary and 2 Button controls on the Controller panel.
Sustain Pedal	This allows you to assign the Sustain Pedal as a modulation source.
Polyphony	This allows you to apply modulation according to how many notes you play. E.g. you could have a short envelope attack when you play single notes, and a long attack when you play chords.
Step Sequencer	This allows you to apply modulation according to the settings for each step in the Step Sequencer. On the sub-menu you can chose to apply modulation according to Gate/Note/Curve 1 and 2/Gate Length/Step Duration settings for each step. In addition you have Start and End Trig, which sends a gate trigger at the start and end of the Step sequence, respectively.
CV Inputs 1-4	These are CV inputs on the back panel which facilitates the use of external modulation sources, (e.g. the Matrix) in Thor. If connected you can freely assign the external CV to any modulation destination in Thor.
Audio Inputs 1-4	These are Audio inputs on the back panel which allows you to connect external audio signals and process these using Thor parameters, or use them as modulation sources. See "About using the Audio inputs".

Modulation Destinations - Voice section

The following parameters can be used as Voice section modulation Destinations:

Parameter	Description
Osc 1	There are four modulation destinations available on the Osc 1 sub-menu: • **Pitch - this will affect oscillator pitch (frequency).** • **FM - this will frequency modulate the oscillator.** The difference between Pitch and FM is that if a high frequency audio signal (i.e. an oscillator or an external audio signal) is the source, FM will not alter the basic pitch of the source, only the timbre. If Pitch is used both the pitch and the timbre will be affected. • **There is also a modifier parameter, which differs depending on what oscillator type is selected. See "The Oscillator section" for details.** • **Osc 2 AM Amount - this will control AM modulation amount from Osc 2. See "About Amplitude Modulation (AM)".**
Osc 2/ Osc 3	Oscillator slots 2 and 3 have the same Destination parameters as Osc 1, except that there is no AM.
Filter 1/ Filter 2	The following destinations are available on the Filter 1 and 2 sub-menus: • **Audio In - this allows you to connect an audio source (e.g. an oscillator or an external audio signal) to the filter input.** • **Frequency - this controls the filter frequency.** • **Frequency (FM) - this will apply filter frequency modulation.** The difference between Frequency and FM is that if a high frequency audio signal (i.e. an oscillator or an external audio signal) is the source, FM will not alter the basic frequency of the source, only the timbre. If Frequency is used both the pitch and the timbre will be affected. • **Resonance - this controls filter resonance.** • **Drive - this controls the filter's Drive parameter.** • **Gender - this controls the Gender parameter (Formant filter only).** • **LPHPMix - this controls the LP/HP parameter (State Variable filter only).**
Shaper Drive	This will control the Shaper Drive parameter.
Amp	The Amp section has three destinations on the sub-menu: • **Input - this allows you to connect a source (e.g. an oscillator or an external audio signal) to the Amp input.** • **Gain - this controls the Amp Gain.** • **Pan - this controls the Pan for each voice. Modulating this parameter with for example LFO 1 means that the Pan position will modulate differently for each voice you play.**
Mix	The Mixer has three destinations on the sub-menu: • **Osc 1+2 Level - this controls the level of both oscillator 1 and 2.** • **Osc 1:2 Balance - you can modulate the level balance between oscillator 1 and 2, e.g. to sweep from one oscillator to the other.** • **Osc 3 Level - this controls the level of oscillator 3.**

Parameter	Description
Filter Envelope	The Filter Envelope mod destinations are as follows: • **Gate - this is the gate input of the envelope. A gate signal applied to this input will trigger the envelope.** • **Attack - this controls the Attack of the envelope.** • **Decay - this controls the Decay of the envelope.** • **Release - this controls the Release parameter.**
Amp Envelope	This has the same destination parameters as the Filter Envelope.
Mod Envelope	This has the same destination parameters as the Filter Envelope.
LFO 1 Rate	This allows you to control the LFO 1 Rate parameter.

Modulation Destinations - Global

The following Global modulation destinations are available:

Parameter	Description
Portamento	This allows you to control the Portamento time parameter.
LFO 2 Rate	This allows you to control the LFO 2 Rate parameter.
Global Envelope	The Global Envelope mod destinations are as follows: • **Gate - this is the gate input of the envelope. A gate signal applied to this input will trigger the envelope.** • **Attack - this controls the attack time of the envelope.** • **Decay - this controls the decay time of the envelope.** • **Release - this controls the release time of the envelope.**
Filter 3	The following destinations are available on the Filter 3 sub-menu: • **Left/Right In - this allows you to connect a source to the filter input.** • **Frequency - this controls the filter frequency.** • **Frequency (FM) - this will apply filter frequency modulation.** • **Resonance - this controls filter resonance.** • **Drive - this controls the filter's Drive parameter.** • **Gender - this controls the Gender parameter (Formant filter only).** • **LPHPMix - this controls the LP/HP parameter (State Variable filter only).**
Chorus	The Chorus effect has the following destinations: • **DryWet balance** • **Delay (time)** • **ModRate** • **ModAmount** • **Feedback**

Parameter	Description
Delay	The Delay effect has the following destinations: • **DryWet balance** • **Time** • **ModRate** • **ModAmount** • **Feedback**
Step Sequencer	This allows you to control various parameters belonging to the Step Sequencer. • **Trig - this enables control over the Step Sequencer Run on/off status.** • **Rate - this enables control over the Step Sequencer Rate.** • **Transpose - this enables control over the Step Sequencer base pitch. E.g. if you apply MIDI Note as a source to this parameter you can transpose the sequence by playing notes.** • **Velocity - this enables control over the Step Sequencer Velocity response.** • **Gate Length - this enables control over the Step Sequencer Gate Length response.**
CV Output 1-4	This will allow you to send signals to the CV outputs on the back of the device. Note that you can send CV signals to audio outputs and vice versa.
Audio Output 1-4	This will allow you to send signals to the audio outputs on the back of the device. Note that you can send CV signals to audio outputs and vice versa.

Scale parameters

The available scale parameters are the same as the Source parameters.

About using the Audio inputs

The 4 Audio inputs on the back panel can be used to connect external audio sources and process them with Thor's parameters.

Note that when routing audio to the Voice section, the following things apply:

- **There are only mono inputs in the Voice section.**
- **You need to send a gate trigger for the audio signal to be heard. This can be done in three ways; by playing notes, via notes played by the Step sequencer or from CV gate signals.**

→ **Routing audio to Global destinations does not require a gate trigger and stereo inputs are provided.**

→ **The external audio sources can also be used purely for modulation, e.g. you can modulate an oscillators pitch with an audio signal.**
This way you can use the audio input source to modulate any available destinations.



Step Sequencer



Thor's Step Sequencer is a further development of the step sequencers which were often present in vintage analog modular systems. It can be used for programming arpeggios or short melody sequences. Alternatively, it can be used purely as a modulation source.

You can have up to 16 steps, and each step can be programmed with various values such as Note pitch, Velocity, Step Duration etc.

Basic operation

The main parameters and functions are as follows:

- **The row of 16 buttons are used to program each step's on or off status.**
 A lit button means that the step is active, and a dark button means that the step will be a rest (silent).
- **Each step button has a knob above it, which is used to set values for the corresponding step.**
- **The Edit knob determines what value you set with the step knobs.**
 The available Edit values are Note (pitch), Velocity, Gate length, Step duration and Curve 1 and 2.
- **The Run button starts/stops the step sequencer.**

What exactly happens when you press Run depends on the Run mode - see below.

Setting the Run mode



The Run mode is set with the lever beside the Run button. The set mode governs how the step sequencer is played back when you press Run. The options are as follows:

- **Repeat mode - this will repeat the sequence continuously.**
 Click the Run button again or use the Transport to stop.
- **1 Shot mode - this will play the sequence once then stop.**
- **Step mode - the Run button steps the sequencer forward one step at a time.**
- **Off - the step sequencer is inactive.**

Setting the direction



The Direction parameter is used to set the direction of the step sequence. The following options are available:

- **Forward - plays the sequence from the first step to the last.**
- **Reverse - plays the sequence from the last step to the first.**
- **Pendulum 1 - plays the sequence from the first step to the last, then from the last step to the first.**
 I.e. the last and first step is played twice when the sequencer reverses direction.
- **Pendulum 2 - plays the sequence from the first step to the last, then from the second last step to the first, i.e. without repeating the last/first step when reversing direction.**
- **Random - plays the steps in a random order.**

Programming step note pitch

To program step note pitch, you proceed as follows:

1. **Make sure that the Step Seq Trigger button is activated in the Controller panel.**



2. **Set the Run mode to “Repeat”.**
 You don’t have to use Repeat mode but it makes it easier to follow the following steps.
3. **Start the step sequencer by pressing the Run button.**
 You should now hear a sequence of repeated notes, each with the same pitch (C3). The current step is indicated by a yellow LED above the step buttons.
4. **Make sure that the Edit knob is set to Note.**



5. **Turn one of the step knobs above one of the steps.**
 A tooltip shows you what current note pitch the knob is set to, and when the sequencer repeats you should be able to hear the change in pitch for that step. Turn clockwise to raise the pitch in semitone increments. Turn counter-clockwise to lower the pitch.



- **You can set the knob's note range by using the Octave lever to the left of the step buttons.**
 Available note ranges are 2 Octaves (i.e. one octave up and down from the middle knob position (C3), 4 Octaves (i.e. two octaves up and down from the middle position (C3), or Full (-C2 to G8).
- **Note that the octave range can be set independently for each step. Each step memorizes the current octave range when the pitch is set for that step, and will keep this octave range until you change the pitch for the step with a different octave range setting.**
- **You can either program steps "on the fly" (with the Step sequencer running) or step by step (Step mode).**
 In Step mode, you press Run to forward the step number one position so you can set step parameters for one step at a time.

By using this general method you can continue to enter note pitch for other steps.

Inserting rests

To make step sequences more rhythmically interesting, you can program rests for steps.

- **This is simply done by pressing one or several step buttons so they go dark.**
 Dark steps will be rests.
- **Note that the Step Duration value still "counts" for rests.**

Setting the number of steps

- **You can set how many steps a sequence should have before starting over using the Steps knob at the far right on the panel.**
 Up to 16 steps can be used. The lit LEDs above each step button show the number of steps currently used. You can also change number of steps by clicking on a LED directly - the sequencer will then stop/start over at the selected step.



Setting Rate

The Rate knob determines the rate of the step sequence.

- **You can either use "free running" rates (i.e. not synced to main sequencer tempo) or synced tempo.**
 This is set with the Sync button on/off status. If Sync is active you can set the tempo in various beat resolutions.

Setting other values for steps

For each step you can also program other parameters with the step value knobs apart from note pitch. You use the Edit knob to set one the following:

- **Velocity - if this is selected as the Edit mode you can set a velocity value for each step.**
 Default value is 100, range is 0-127.
- **Gate Length - if this is selected as the Edit mode you can set a Gate Length value for each step.**
 Default is 75%. Gate Length determines the length of the note played for that step.
- **Step Duration - if this is selected as the Edit mode you can set a Step duration value for each step.**
 This parameter determines the total length of the step, which is a factor related to the sequencer rate. Range is 1/4 to 4. E.g. if Rate is 1/16, "1" means a 1/16-note will be played, a "4" means a 1/4-note will be played, and so on.
- **The Curves 1 and 2 allow you to set values for each step that can be sent to control parameters of your choice.**
 This is done in the Modulation bus routing section, where these two independent Curves are selectable as Source controllers.
 - **You can compare these curves to the Curve CV output of the Matrix - they simply represent a series of values which can be applied to anything.**

Step Pattern functions

You will find some specific Step pattern functions on the Edit menu (and on the device context menu). These are as follows:

Function	Description
Randomize Sequencer Pattern	The Randomize Pattern function creates random patterns. The function only randomizes the selected Edit value (e.g. if set to Note, only the note pitch values are randomized, not velocity, gate length etc.).
Shift Pattern L/R	The Shift Pattern functions move the pattern one step to the left or right. All parameters (rests, note pitch, velocity etc.) are shifted one step.

Connections

The following Audio and CV connectors can be found at the back of Thor:

Sequencer Control Inputs

The Sequencer Control CV and Gate inputs allow you to play Thor from another CV/Gate device (e.g. a Matrix or the RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity.

Modulation Inputs

- **The Rotary control voltage (CV) inputs (with associated voltage trim pots), can modulate the two virtual Rotary controls.**
 Thus, any parameter(s) assigned to a Rotary control can be modulated by CV.
- **The Filter 1x allows for CV control of the Filter 1 frequency.**
 If the Formant filter is used this is the "X" parameter - see "Formant filter".
- **The four CV Inputs can receive CV from external sources that will be available as Sources in the Modulation bus.**

Modulation Outputs

Here you can find CV outputs from the Global Envelope and LFO 2, as well as the 4 user assignable CV outputs.

Audio Inputs

The Audio inputs can be used to connect audio outputs from other Reason devices. When connected, you can route the Audio inputs as a Modulation source to for example one of the filters and process the external signal. See "About using the Audio inputs".

Audio Outputs

Thor has 4 outputs:

- **1 Left (Mono)/2 Right - these are the main stereo outputs.**
- **2 additional outputs (3 and 4), which can be assigned in the Modulation section.**



Chapter 17
Subtractor Synthesizer

Introduction

Subtractor is an analog-type polyphonic synthesizer based on subtractive synthesis, the method used in analog synthesizers. This chapter will go through all parameters of each section of Subtractor. In addition to the parameter descriptions, the chapter also includes a few tips and tricks to help you get the most out of the Subtractor synthesizer.

! **It is recommended that you start with default settings (an "Init Patch") if you intend to follow the examples in this chapter, unless otherwise is stated. An Init Patch is created by selecting "Reset Device" from the device's context menu, or from the Edit menu. If you wish to keep the current settings, save them before initializing.**

The Subtractor has the following basic features:

- **Up to 99 Voice Polyphony.**
 You can set the number of voices for each Patch.
- **Dual Filters.**
 A combination of a multimode filter and a second, linkable, lowpass filter allows for complex filtering effects. See "The Filter Section".
- **Two Oscillators, each with 32 waveforms.**
 See "The Oscillator Section".
- **Frequency Modulation (FM).**
 See "Frequency Modulation (FM)".
- **Oscillator Phase Offset Modulation.**
 This is an unique Subtractor feature that generates waveform variations. See "Phase Offset Modulation".
- **Two Low Frequency Oscillators (LFO's)**
 See "LFO Section".
- **Three Envelope Generators.**
 See "Envelopes - General".
- **Extensive Velocity Control.**
 See "Velocity Control".
- **Extensive CV/Gate Modulation possibilities.**
 See "Connections".

Loading and Saving Patches

Loading and saving patches is done in the same way as with any other Reason device - see "Loading patches" and "Saving patches".

The Oscillator Section



Subtractor provides two oscillators. Oscillators are the main sound generators in Subtractor, the other features are used to shape the sound of the oscillators. Oscillators generate two basic properties, waveform and pitch (frequency). The type of waveform the oscillator produces determines the harmonic content of the sound, which in turn affects the resultant sound quality (timbre). Selecting a oscillator waveform is usually the starting point when creating a new Subtractor Patch from scratch.

Oscillator 1 Waveform



Oscillator 1 provides 32 waveforms. The first four are standard waveforms, and the rest are “special” waveforms, some of which are suitable for emulating various musical instrument sounds.

- **It is worth noting here that all waveforms can be radically transformed using Phase offset modulation (see “Phase Offset Modulation”).**

→ **To select a waveform, click the spin controls to the right of the “Waveform” LED display.**
The first 4 basic waveforms are shown as standard symbols, and the special waveforms are numbered 5 - 32.

Here follows a brief description of the Subtractor waveforms:

! **Please note that the descriptions of the waveforms sound or timbre is merely meant to provide a basic guideline, and shouldn't be taken too literally. Given the myriad ways you can modulate and distort a waveform in Subtractor, you can produce extremely different results from any given waveform.**

Waveform	Description
Sawtooth 	This waveform contains all harmonics and produces a bright and rich sound. The Sawtooth is perhaps the most "general purpose" of all the available waveforms.
Square 	A square wave only contains odd number harmonics, which produces a distinct, hollow sound.
Triangle 	The Triangle waveform generates only a few harmonics, spaced at odd harmonic numbers. This produces a flute-like sound, with a slightly hollow character.
Sine 	The sine wave is the simplest possible waveform, with no harmonics (overtones). The sine wave produces a neutral, soft timbre.
5	This waveform emphasizes the higher harmonics, a bit like a sawtooth wave, only slightly less bright-sounding.
6	This waveform features a rich, complex harmonic structure, suitable for emulating the sound of an acoustic piano.
7	This waveform generates a glassy, smooth timbre. Good for electric piano-type sounds.
8	This waveform is suitable for keyboard-type sounds such as harpsichord or clavinet.
9	This waveform is suitable for electric bass-type sounds.
10	This is a good waveform for deep, sub-bass sounds.
11	This produces a waveform with strong formants, suitable for voice-like sounds.
12	This waveform produces a metallic timbre, suitable for a variety of sounds.
13	This produces a waveform suitable for organ-type sounds.
14	This waveform is also good for organ-type sounds. Has a brighter sound compared to waveform 13.
15	This waveform is suitable for bowed string sounds, like violin or cello.
16	Similar to 15, but with a slightly different character.
17	Another waveform suitable for string-type sounds.
18	This waveform is rich in harmonics and suitable for steel string guitar-type sounds.
19	This waveform is suitable for brass-type sounds.
20	This waveform is suitable for muted brass-type sounds.
21	This waveform is suitable for saxophone-like sounds.
22	A waveform suitable for brass and trumpet-type sounds.
23	This waveform is good for emulating mallet instruments such as marimba.
24	Similar to 23, but with a slightly different character.
25	This waveform is suitable for guitar-type sounds.
26	This is a good waveform for plucked string sounds, like harp.
27	Another waveform suitable for mallet-type sounds (see 23-24), but has a brighter quality, good for vibraphone-type sounds.

Waveform	Description
28	Similar to 27, but with a slightly different character.
29	This waveform has complex, enharmonic overtones, suitable for metallic bell-type sounds.
30	Similar to 29, but with a slightly different character. By using FM and setting the Osc Mix to Osc 1, this and the following two waveforms can produce noise.
31	Similar to 30, but with a slightly different character.
32	Similar to 30, but with a slightly different character.

Setting Oscillator 1 Frequency - Octave/Semitone/Cent



By clicking the corresponding up/down buttons you can tune, i.e. change the frequency of Oscillator 1 in three ways:

- **In Octave steps**
 The range is 0 - 9. The default setting is 4 (where "A" above middle "C" on your keyboard generates 440 Hz).
- **In Semitone steps**
 Allows you to raise the frequency in 12 semitone steps (1 octave).
- **In Cent steps (100th of a semitone)**
 The range is -50 to 50 (down or up half a semitone).

Oscillator Keyboard Tracking



Oscillator 1 has a button named "Kbd. Track". If this is switched off, the oscillator pitch will remain constant, regardless of any incoming note pitch messages, although the oscillator still reacts to note on/off messages. This can be useful for certain applications:

- **When Frequency Modulation (FM - see "Frequency Modulation (FM)") or Ring Modulation (see "Ring Modulation") is used.**
 This produces enharmonic sounds with very varying timbre across the keyboard.
- **For special effects and non-pitched sounds (like drums or percussion) that should sound the same across the keyboard.**

Using Oscillator 2



You activate Osc 2 by clicking the button next to the text "Osc 2". Setting oscillator frequency and keyboard tracking is identical to Oscillator 1.

Adding a second oscillator enables many new modulation possibilities which can produce richer timbres. A basic example is to slightly detune (+/– a few cents) one of the oscillators. This slight frequency offset causes the oscillators to "beat" against each other, producing a wider and richer sound. Also, by combining two different waveforms, and adding frequency or ring modulation, many new timbres can be created.

Oscillator Mix



The Osc Mix knob determines the output balance between Osc 1 and Osc 2. To be able to clearly hear both oscillators, the "Osc Mix" knob should be set somewhere around the center position. If you turn the Mix knob fully to the left, only Osc 1 will be heard, and vice versa. [Command]/[Ctrl]-clicking the knob sets the Mix parameter to center position.

Oscillator 2 Waveform

The waveform alternatives for Oscillator 2 are identical to those of Oscillator 1.

However, the Noise Generator provides a third sound generating source (in addition to the two oscillators) in Subtractor, and could be regarded as an "extra" waveform for Oscillator 2, as it is internally routed to the Oscillator 2 output. See below for a description of the Noise Generator.

Noise Generator



The Noise Generator could be viewed as an oscillator that produces noise instead of a pitched waveform. Noise can be used to produce a variety of sounds, the classic example being "wind" or "rolling wave" sounds, where noise is passed through a filter while modulating the filter frequency. Other common applications include non-pitched sounds like drums and percussion, or simulating breath noises for wind instruments. To use the Noise Generator, select an Init Patch and proceed as follows:

1. **Turn Osc 2 off.**
2. **Click the button (in the Noise Generator section) to activate the Noise Generator.**
 If you play a few notes on your MIDI instrument you should now hear Osc1 mixed with the sound of the Noise Generator.
3. **Turn the Mix knob fully to the right, and play a few more notes.**
 Now just the Noise Generator will be heard.

→ **Thus, the output of the Noise Generator is internally routed to Osc 2.**
If you switch Osc 2 on, the noise will be mixed with the Osc 2 waveform.

There are three Noise Generator parameters. These are as follows:

Parameter	Description
Noise Decay	This controls how long it takes for the noise to fade out when you play a note. Note that this is independent from the Amp Envelope Decay parameter, allowing you to mix a short "burst" of noise at the very beginning of a sound, i.e. a pitched sound that uses oscillators together with noise.
Noise Color	This parameter allows you to vary the character of the noise. If the knob is turned fully clockwise, pure or "white" noise (where all frequencies are represented with equal energy) is generated. Turning the knob anti-clockwise produces a gradually less bright sounding noise. Fully anti-clockwise the noise produced is an earthquake-like low frequency rumble.
Level	Controls the level of the Noise Generator.

Phase Offset Modulation

A unique feature of the Subtractor oscillators is the ability to create an extra waveform within one oscillator, to offset the *phase* of that extra waveform, and to modulate this phase offset. By subtracting or multiplying a waveform with a phase offset copy of itself, very complex waveforms can be created. Sounds complicated? Well, the theory behind it might be, but from a user perspective it is just a method of generating new waveforms from existing waveforms.

A seasoned synth programmer using Subtractor for the first time may wonder why the Subtractor oscillators (seemingly) cannot provide the commonly used pulse waveform and the associated pulse width modulation (PWM). Or oscillator sync, another common feature in analog synthesizers. The simple answer is that Subtractor can easily create pulse waveforms (with PWM) and oscillator sync-sounds, and a lot more besides, partly by the use of phase offset modulation.

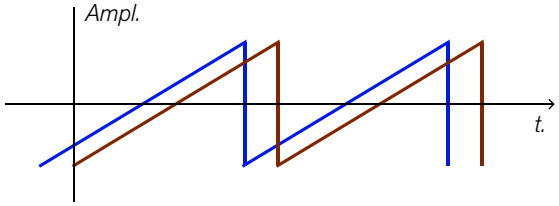


Each oscillator has it's own Phase knob and a selector button. The Phase knob is used to set the amount of phase offset, and the selector switches between three modes:

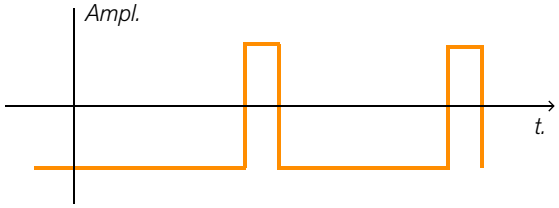
- **Waveform multiplication (x)**
- **Waveform subtraction (–)**
- **No phase offset modulation (o).**

When phase offset modulation is activated, the oscillator creates a *second* waveform of the same shape and offsets it by the amount set with the Phase knob. Depending on the selected mode, Subtractor then either subtracts or multiplies the two waveforms with each other. The resulting waveforms can be seen in the illustration below.

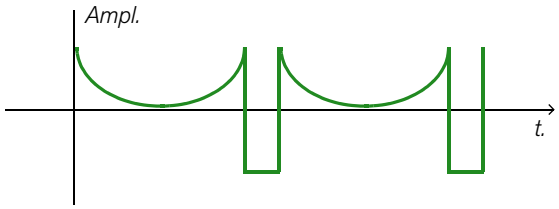
1. The two offset waveforms:



2. The result of subtraction:



3. The result of multiplication:



- **In example 1, we see two sawtooth waves with a slight offset.**
- **Example 2 shows that subtracting one slightly offset sawtooth wave from the other, produces a pulse wave. If you modulate the Phase offset parameter (with for example an LFO), the result will be pulse width modulation (PWM).**
- **Example 3 shows the resulting waveform when multiplying the offset waves with each other. As you can see (and hear if you try it), multiplying waveforms can produce very dramatic and sometimes unexpected results.**

Using phase offset modulation can create very rich and varied timbres, especially when used along with LFO or Envelopes to modulate the phase offset.

► **To get a "feel" for this concept, you could study Patches that use phase offset modulation, and maybe tweak some of the Phase Offset parameters to find out what happens. Try "SyncedUp" in the Polysynth category in the Factory Soundbank for an example of osc sync or "Sweeping Strings" (in the Pads category) for an example of PWM.**

! **Note that if you activate waveform subtraction with a Phase offset set to "0" for an oscillator, the second waveform will cancel out the original waveform completely, and the oscillator output will be silent. If you set the Phase Offset knob to any other value than zero, the sound returns.**

Frequency Modulation (FM)



In synthesizer-speak, Frequency Modulation, or FM, is when the frequency of one oscillator (called the "carrier") is modulated by the frequency of another oscillator (called the "modulator"). Using FM can produce a wide range of harmonic and non harmonic sounds. In Subtractor, Osc 1 is the carrier and Osc 2 the modulator. To try out some of the effects FM can produce, proceed as follows:

1. **Select an Init Patch by selecting "Initialize Patch" from the Edit menu.**
2. **Activate Osc 2.**
 As you need both a carrier and a modulator to produce FM, turning the FM knob will not produce any effect unless you first activate Osc 2. For classic FM sounds, use sine wave on oscillator 1 and triangle wave on oscillator 2.
3. **Use the FM knob to set the FM amount to a value of about 50.**
 As you can hear, the timbre changes, but the effect isn't very pronounced yet.
4. **Turn the Osc Mix knob fully to the left, so that only the sound of Osc 1 is heard.**
 The modulator (Osc 2) still affects Osc 1, even though the Osc 2 output is muted.
5. **Now, hold down a note on your MIDI keyboard and tune Osc 2 a fifth up from the original pitch by setting the Osc 2 frequency "Semi" parameter to a value of 7.**
 As you can hear, for each semitone step you vary the Osc 2 frequency, the timbre changes dramatically. Setting Osc 2 frequency to certain musical intervals (i.e. fourth, fifth or octave semitone steps) produces harmonic, rich timbres, almost like tube distortion. Setting Osc 2 to non-musical intervals usually results in complex, enharmonic timbres.

- **Experiment with different oscillator parameters such as phase offset modulation, changing the waveforms etc. and listen to how they affect the sound of frequency modulation.**

Using the Noise Generator as the Modulator source

As explained earlier, the Noise Generator is internally routed to the Osc 2 output. Hence, if you deactivate Osc 2, and activate the Noise Generator while using FM, the noise will be used to frequency modulate Osc 1.

- **With the Noise Generators default settings, this will sound much like colored noise. But by changing (lowering) the Noise Generator Decay parameter, so that the noise modulates only the attack portion of the sound can produce more interesting results. You could also use a combination of noise and Osc 2.**

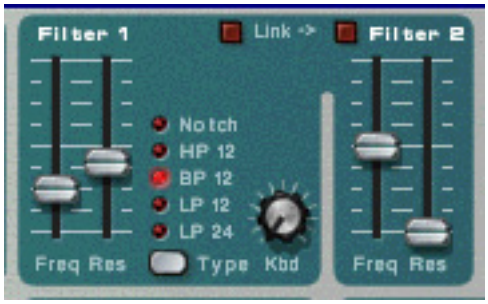
Ring Modulation



Ring Modulators basically multiply two audio signals together. The ring modulated output contains added frequencies generated by the sum of, and the difference between, the frequencies of the two signals. In the Subtractor Ring Modulator, Osc 1 is multiplied with Osc 2 to produce sum and difference frequencies. Ring modulation can be used to create complex and enharmonic, bell-like sounds.

1. **Select an Init Patch by selecting "Initialize Patch" from the Edit menu.**
 Save any current settings you wish to keep before initializing.
2. **Activate Ring Modulation with the button in the lower right corner of the oscillator section.**
3. **Activate Osc 2.**
 You need to activate Osc 2 before any ring modulation can happen.
4. **Turn the Osc Mix knob fully to the right, so that only the sound of Osc 2 is heard.**
 Osc 2 provides the ring modulated output.
5. **If you play a few notes while varying the frequency of either oscillator, by using the Semitone spin controls, you can hear that the timbre changes dramatically.**
 If the oscillators are tuned to the same frequency, and no modulation is applied to either the Osc 1 or 2 frequency, the Ring Modulator won't do much. It is when the frequencies of Osc 1 and Osc 2 differ, that you get the "true" sound of ring modulation.

The Filter Section

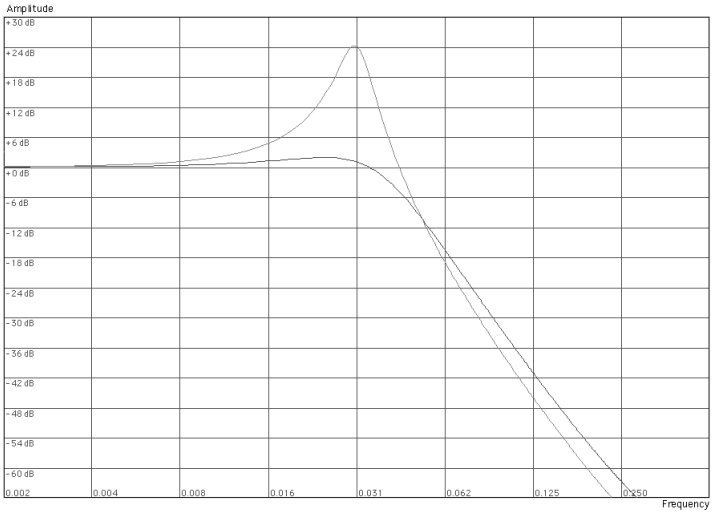


In subtractive synthesis, a filter is the most important tool for shaping the overall timbre of the sound. The filter section in Subtractor contains two filters, the first being a multimode filter with five filter types, and the second being a low-pass filter. The combination of a multimode filter and a lowpass filter can be used to create very complex filter effects.

Filter 1 Type

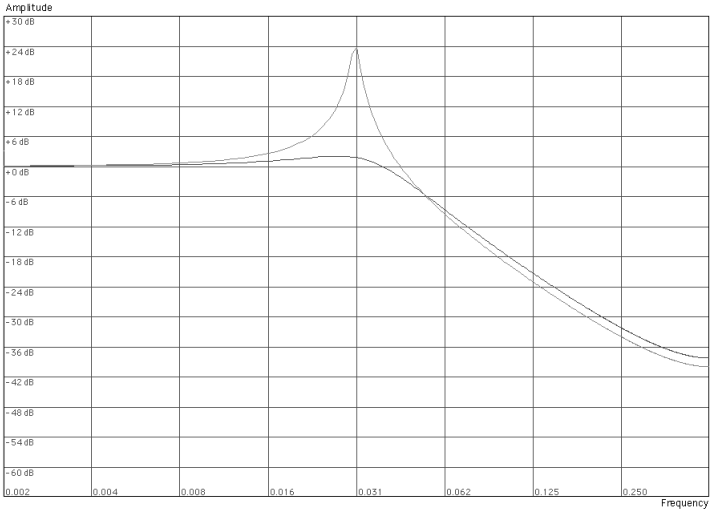
With this multi-selector you can set Filter 1 to operate as one of five different types of filter. The five types are illustrated and explained on the following pages:

- **24 dB Lowpass (LP 24)**
 Lowpass filters lets low frequencies pass and cuts out the high frequencies. This filter type has a fairly steep roll-off curve (24dB/Octave). Many classic synthesizers (Minimoog/Prophet 5 etc.) use this filter type.



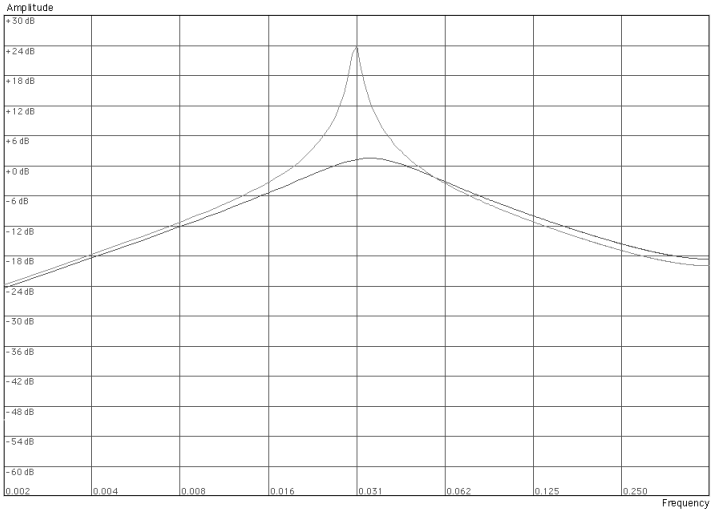
The darker curve illustrates the roll-off curve of the 24dB Lowpass Filter. The lighter curve in the middle represents the filter characteristic when the Resonance parameter is raised.

- **12 dB Lowpass (LP 12)**
 This type of lowpass filter is also widely used in analog synthesizers (Oberheim, early Korg synths etc.). It has a gentler slope (12 dB/Octave), leaving more of the harmonics in the filtered sound compared to the LP 24 filter.



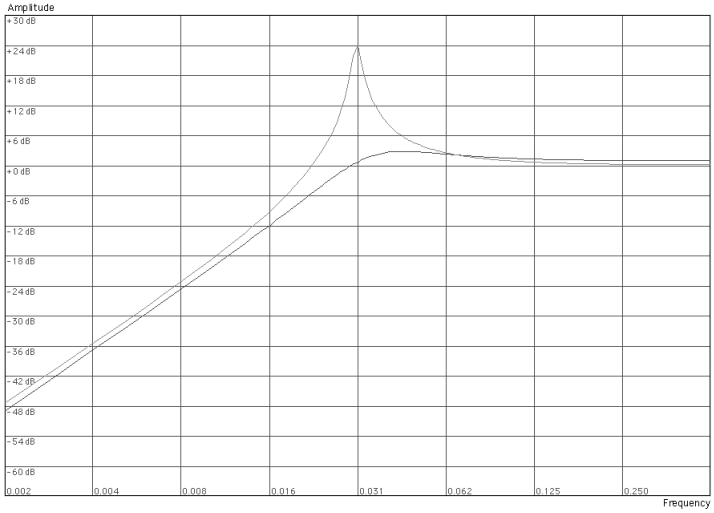
The darker curve illustrates the roll-off curve of the 12dB Lowpass Filter. The lighter curve in the middle represents the filter characteristic when the Resonance parameter is raised.

- **Bandpass (BP 12)**
 A bandpass filter cuts both high and low frequencies, while midrange frequencies are not affected. Each slope in this filter type has a 12 dB/Octave roll-off.



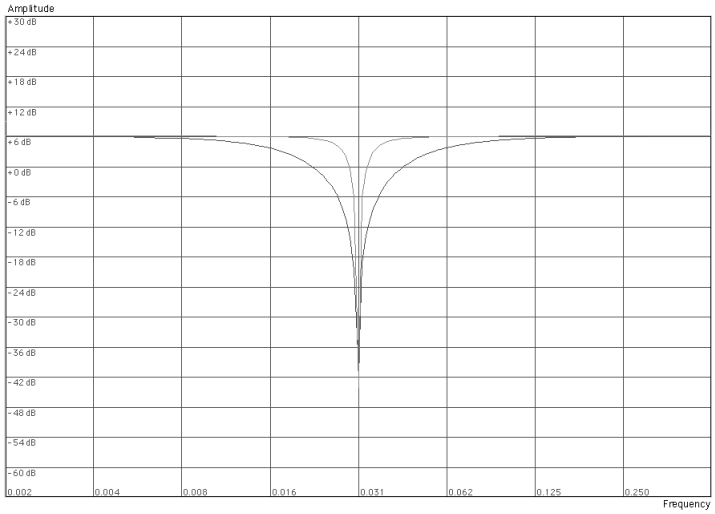
The darker curve illustrates the roll-off curve of the Bandpass Filter. The lighter curve in the middle represents the filter characteristic when the Resonance parameter is raised.

- **Highpass (HP12)**
 A highpass filter is the opposite of a lowpass filter, cutting out lower frequencies and letting high frequencies pass. The HP filter slope has a 12 dB/Octave roll-off.



The darker curve illustrates the roll-off curve of the Highpass Filter. The lighter curve in the middle represents the filter characteristic when the Resonance parameter is raised.

- **Notch**
 A notch filter (or band reject filter) could be described as the opposite of a bandpass filter. It cuts off frequencies in a narrow midrange band, letting the frequencies below and above through. On its own, a notch filter doesn't really alter the timbre in any dramatic way, simply because most frequencies are let through. However, by combining a notch filter with a lowpass filter (using Filter 2 - see "Filter 2"), more musically useful filter characteristics can be created. Such a filter combination can produce soft timbres that still sound "clear". The effect is especially noticeable with low resonance (see "Resonance") settings.



The darker curve illustrates the roll-off curve of the Notch Filter. The lighter curve in the middle represents the filter characteristic when the Resonance parameter is raised.

Filter 1 Frequency

The Filter Frequency parameter (often referred to as “cutoff”) determines which area of the frequency spectrum the filter will operate in. For a lowpass filter, the frequency parameter could be described as governing the “opening” and “closing” of the filter. If the Filter Freq is set to zero, none or only the very lowest frequencies are heard, if set to maximum, all frequencies in the waveform are heard. Gradually changing the Filter Frequency produces the classic synthesizer filter “sweep” sound.

! **Note that the Filter Frequency parameter is usually controlled by the Filter Envelope (see “Filter Envelope”) as well. Changing the Filter Frequency with the Freq slider may therefore not produce the expected result.**

Resonance

The filter resonance parameter is used to set the Filter characteristic, or quality. For lowpass filters, raising the filter Res value will emphasize the frequencies around the set filter frequency. This produces a generally thinner sound, but with a sharper, more pronounced filter frequency “sweep”. The higher the filter Res value, the more resonant the sound becomes until it produces a whistling or ringing sound. If you set a high value for the Res parameter and then vary the filter frequency, this will produce a very distinct sweep, with the ringing sound being very evident at certain frequencies.

- **For the highpass filter, the Res parameter operates just like for the lowpass filters.**
- **When you use the Bandpass or Notch filter, the Resonance setting adjusts the width of the band. When you raise the Resonance, the band where frequencies are let through (Bandpass), or cut (Notch) will become narrower. Generally, the Notch filter produces more musical results using low resonance settings.**

Filter Keyboard Track (Kbd)

If Filter Keyboard Track is activated, the filter frequency will increase the further up on the keyboard you play. If a lowpass filter frequency is constant (a Kbd setting of “0”) this can introduce a certain loss of “sparkle” in a sound the higher up the keyboard you play, because the harmonics in the sound are progressively being cut. By using a degree of Filter Keyboard Tracking, this can be compensated for.

Filter 2

A very useful and unusual feature of the Subtractor Synthesizer is the presence of an additional 12dB/Oct lowpass filter. Using two filters together can produce many interesting filter characteristics, that would be impossible to create using a single filter, for example formant effects.

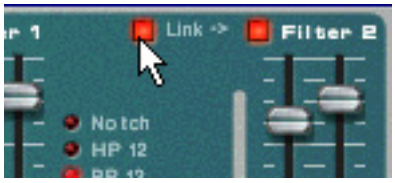
The parameters are identical to Filter 1, except in that the filter type is fixed, and it does not have filter keyboard tracking.

→ **To activate Filter 2, click the button at the top of the Filter 2 section.**
Filter 1 and Filter 2 are connected in series. This means that the output of Filter 1 is routed to Filter 2, but both filters function independently. For example, if Filter 1 was filtering out most of the frequencies, this would leave Filter 2 very little to "work with". Similarly, if Filter 2 had a filter frequency setting of "0", all frequencies would be filtered out regardless of the settings of Filter 1.



► **Try the "Singing Synth" patch (in the Monosynth category of the Factory Sound Bank) for an example of how dual filters can be used.**

Filter Link



When Link (and Filter 2) is activated, the Filter 1 frequency controls the frequency offset of Filter 2. That is, if you have set different filter frequency values for Filter 1 and 2, changing the Filter 1 frequency will also change the frequency for Filter 2, but keeping the relative offset.

► **Try the "Fozzy Fonk" patch (in the Polysynth category of the Factory Sound Bank) for an example how linked filters can be used.**

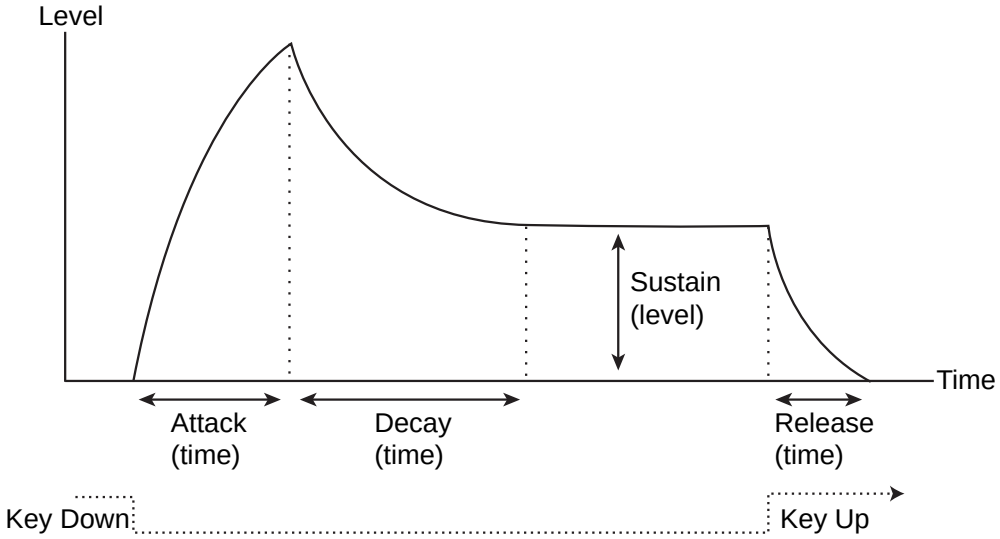
! **Caution! If no filter modulation is used, and the filters are linked, pulling down the frequency of Filter 2 to zero will cause both filters to be set to the same frequency. If combined with high Res settings, this can produce *very* loud volume levels that cause distortion!**

Envelopes - General

Envelope generators are used to control several important sound parameters in analog synthesizers, such as pitch, volume, filter frequency etc. Envelopes govern how these parameters should respond over time - from the moment a note is struck to the moment it is released.

Standard synthesizer envelope generators have four parameters; Attack, Decay, Sustain and Release (ADSR).

There are three envelope generators in the Subtractor, one for volume, one for the Filter 1 frequency, and one modulation envelope which has selectable modulation destinations.



The ADSR envelope parameters.

Attack

When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the maximum value. How long this should take, depends on the Attack setting. If the Attack is set to "0", the maximum value is reached instantly. If this value is raised, it will take time before the maximum value is reached.

For example, if the Attack value is raised and the envelope is controlling the filter frequency, the filter frequency will gradually rise up to a point each time a key is pressed, like an "auto-wha" effect.

Decay

After the maximum value has been reached, the value starts to drop. How long this should take is governed by the Decay parameter.

If you wanted to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0" and the Decay parameter should be set to a medium value, so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you use the Sustain parameter.

Sustain

The Sustain parameter determines the level the envelope should rest at, after the Decay. If you set Sustain to full level, the Decay setting is of no importance since the volume of the sound is never lowered.

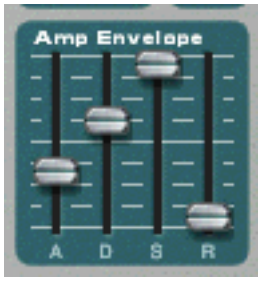
If you wanted to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

But often a combination of Decay and Sustain is used to generate envelopes that rise up to the maximum value, then gradually decreases to finally land to rest on a level somewhere in-between zero and maximum. Note that Sustain represents a *level*, whereas the other envelope parameters represent times.

Release

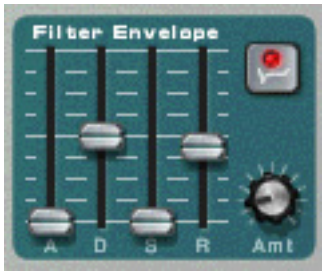
Finally, we have the Release parameter. This works just like the Decay parameter, except it determines the time it takes for the value to fall back to zero *after* releasing the key.

Amplitude Envelope



The Amplitude Envelope is used to adjust how the volume of the sound should change from the time you press a key until the key is released. By setting up a volume envelope you sculpt the sound's basic shape with the four Amplitude Envelope parameters, Attack, Decay, Sustain and Release. This determines the basic character of the sound (soft, long, short etc.).

Filter Envelope



The Filter Envelope affects the Filter 1 Frequency parameter. By setting up a filter envelope you control the how the filter frequency should change over time with the four Filter Envelope parameters, Attack, Decay, Sustain and Release.

Filter Envelope Amount

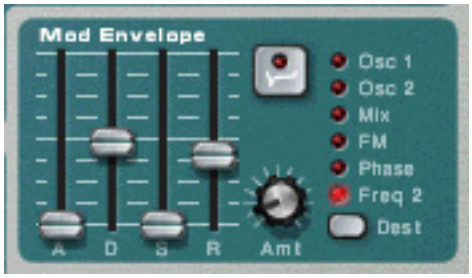
This parameter determines to what degree the filter will be affected by the Filter Envelope. Raising this knob's value creates more drastic results. The Envelope Amount parameter and the set Filter Frequency are related. If the Filter Freq slider is set to around the middle, this means that the moment you press a key the filter is already halfway open. The set Filter Envelope will then open the filter further from this point. The Filter Envelope Amount setting affects how much further the filter will open.

Filter Envelope Invert



If this button is activated, the envelope will be inverted. For example, normally the Decay parameter lowers the filter frequency, but after activating Invert it will instead raise it, by the same amount.

Mod Envelope



The Mod Envelope allows you to select one of a number of parameters, or Destinations, to control with the envelope. By setting up a modulation envelope you control the how the selected Destination parameter should change over time with the four Mod Envelope parameters, Attack, Decay, Sustain and Release.

The available Mod Envelope Destinations are as follows:

Destination	Description
Osc 1	Selecting this makes the Mod Envelope control the pitch (frequency) of Osc 1.
Osc 2	Same as above, but for Osc 2.
Osc Mix	Selecting this makes the Mod Envelope control the oscillator Mix parameter. Both oscillators must be activated for this to have any effect.
FM	Selecting this makes the Mod Envelope control the FM Amount parameter. Both oscillators must be activated for this to have any effect.
Phase	Selecting this makes the Mod Envelope control the Phase Offset parameter for both Osc 1 and 2. Note that Phase Offset Modulation (Subtraction or Multiplication) must be activated for this to have any effect.
Freq 2	Selecting this makes the Mod Envelope control the Frequency parameter for Filter 2.

LFO Section



LFO stands for Low Frequency Oscillator. LFO's are oscillators, just like Osc 1 & 2, in that they also generate a waveform and a frequency. However, there are two significant differences:

- **LFOs only generate waveforms with low frequencies.**
- **The output of the two LFO's are never actually heard. Instead they are used for modulating various parameters.**

The most typical application of an LFO is to modulate the pitch of a (sound generating) oscillator, to produce vibrato. Subtractor is equipped with two LFO's. The parameters and the possible modulation destinations vary somewhat between LFO 1 and LFO 2.

LFO 1 Parameters

Waveform

LFO 1 allows you to select different waveforms for modulating parameters. These are (from top to bottom):

Waveform	Description
Triangle	This is a smooth waveform, suitable for normal vibrato.
Inverted Sawtooth	This produces a "ramp up" cycle. If applied to an oscillator's frequency, the pitch would sweep up to a set point (governed by the Amount setting), after which the cycle immediately starts over.
Sawtooth	This produces a "ramp down" cycle, the same as above but inverted.
Square	This produces cycles that abruptly changes between two values, usable for trills etc.
Random	Produces random stepped modulation to the destination. On some vintage synths, this is called "sample & hold".
Soft Random	The same as above, but with smooth modulation.

Destination

The available LFO 1 Destinations are as follows:

Destination	Description
Osc 1&2	Selecting this makes LFO 1 control the pitch (frequency) of Osc 1 and Osc 2.
Osc 2	Same as above, but for Osc 2.
Filter Freq	Selecting this makes the LFO 1 control the filter frequency for Filter 1 (and Filter 2 if linked).
FM	Selecting this makes the LFO 1 control the FM Amount parameter. Both oscillators must be activated for this to have any effect.
Phase	Selecting this makes the LFO 1 control the Phase Offset parameter for both Osc 1 and 2. Note that Phase Offset Modulation (Subtraction or Multiplication) must be activated for this to have any effect.
Osc Mix	Selecting this makes the LFO 1 control the oscillator Mix parameter.

Sync

By clicking this button you activate/deactivate LFO sync. The frequency of the LFO will then be synchronized to the song tempo, in one of 16 possible time divisions. When sync is activated, the Rate knob (see below) is used for setting the desired time division.

Turn the knob and check the tooltip for an indication of the time division.



Rate

The Rate knob controls the LFO's frequency. Turn clockwise for a faster modulation rate.

Amount

This parameter determines to what degree the selected parameter destination will be affected by LFO 1. Raising this knob's value creates more drastic results.

LFO 2 Parameters

LFO 2 is polyphonic. This means that for every note you play, an *independent* LFO cycle is generated, whereas LFO 1 always modulates the destination parameter using the same "cycle". This can be used to produce subtle cross-modulation effects, with several LFO cycles that "beat" against each other. This also enables LFO 2 to produce modulation rates that vary across the keyboard (see the "Keyboard Tracking" parameter below).

Destination

The available LFO 2 Destinations are as follows:

Destination	Description
Osc 1&2	Selecting this makes LFO 2 modulate the pitch (frequency) of Osc 1 and Osc 2.
Phase	Selecting this makes the LFO 2 modulate the Phase Offset parameter for both Osc 1 and 2. Note that Phase Offset Modulation (Subtraction or Multiplication) must be activated for this to have any effect.
Filter Freq 2	Selecting this makes the LFO 2 modulate the filter frequency for Filter 2.
Amp	Selecting this makes the LFO 2 modulate the overall volume., to create tremolo-effects.

LFO 2 Delay

This parameter is used to set a delay between when a note is played and when the LFO modulation "kicks in". For example, if Osc 1 & 2 is selected as the destination parameter and Delay was set to a moderate value, the sound would start out unmodulated, with the vibrato only setting in if you hold the note(s) long enough. Delayed LFO modulation can be very useful, especially if you are playing musical instrument-like sounds like violin or flute. Naturally it could also be used to control more extreme modulation effects and still retain the "playability" of the sound.

LFO 2 Keyboard Tracking

If LFO keyboard tracking is activated, the LFO rate will progressively increase the higher up on the keyboard you play. Raising this value creates more drastic results.

► **If the LFO is set to modulate the phase offset, LFO keyboard tracking can produce good results. For example, synth string pads and other sounds that use PWM (see "Phase Offset Modulation") can benefit from this.**

Rate

The Rate knob controls the LFO's frequency. Turn clockwise for a faster modulation rate.

Amount

This parameter determines to what degree the selected parameter destination will be affected by LFO 2. Raising this knob's value creates more drastic results.

Play Parameters

This section deals with two things: Parameters that are affected by how you play, and modulation that can be applied manually with standard MIDI keyboard controls.

These are:

- **Velocity Control**
- **Pitch Bend and Modulation Wheel**
- **Legato**
- **Portamento**
- **Polyphony**

Velocity Control



Velocity is used to control various parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder. Subtractor features very comprehensive velocity modulation capabilities. By using the knobs in this section, you can control how much the various parameters will be affected by velocity. The velocity sensitivity amount can be set to either positive or negative values, with the center position representing no velocity control.

The following parameters can be velocity controlled:

Destination	Description
Amp	This let's you velocity control the overall volume of the sound. If a positive value is set, the volume will increase the harder you strike a key. A negative value inverts this relationship, so that the volume decreases if you play harder, and increases if you play softer. If set to zero, the sound will play at a constant volume, regardless of how hard or soft you play.
FM	This sets velocity control for the FM Amount parameter. A positive value will increase the FM amount the harder you play. Negative values invert this relationship.
M. Env	This sets velocity control for the Mod Envelope Amount parameter. A positive value will increase the envelope amount the harder you play. Negative values invert this relationship.
Phase	This sets velocity control for the Phase Offset parameter. This applies to both Osc 1 & 2, but the relative offset values are retained. A positive value will increase the phase offset the harder you play. Negative values invert this relationship.
Freq 2	This sets velocity control for the Filter 2 Frequency parameter. A positive value will increase the filter frequency the harder you play. Negative values invert this relationship.
F. Env	This sets velocity control for the Filter Envelope Amount parameter. A positive value will increase the envelope amount the harder you play. Negative values invert this relationship.
F. Dec	This sets velocity control for the Filter Envelope Decay parameter. A positive value will increase the Decay time the harder you play. Negative values invert this relationship.
Osc Mix	This sets velocity control for the Osc Mix parameter. A positive value will increase the Osc 2 Mix amount the harder you play. Negative values invert this relationship.
A. Attack	This sets velocity control for the Amp Envelope Attack parameter. A positive value will increase the Attack time the harder you play. Negative values invert this relationship.

Pitch Bend and Modulation Wheels



The Pitch Bend wheel is used for "bending" notes, like bending the strings on a guitar. The Modulation wheel can be used to apply various modulation while you are playing. Virtually all MIDI keyboards have Pitch Bend and Modulation controls. Subtractor features not only the settings for how incoming MIDI Pitch Bend and Modulation wheel messages should affect the sound. Subtractor also has two functional wheels that could be used to apply real time modulation and pitch bend should you not have these controllers on your keyboard, or if you aren't using a keyboard at all. The Subtractor wheels mirror the movements of the MIDI keyboard controllers.

Pitch Bend Range

The Range parameter sets the amount of pitch bend when the wheel is turned fully up or down. The maximum range is "24" (=up/down 2 Octaves).

Modulation Wheel

The Modulation wheel can be set to simultaneously control a number of parameters. You can set positive or negative values, just like in the Velocity Control section. The following parameters can be affected by the modulation wheel:

Parameter	Description
F. Freq	This sets modulation wheel control of the Filter 1 Frequency parameter. A positive value will increase the frequency if the wheel is pushed forward. Negative values invert this relationship.
F. Res	This sets modulation wheel control of the Filter 1 Resonance parameter. A positive value will increase the resonance if the wheel is pushed forward. Negative values invert this relationship.
LFO 1	This sets modulation wheel control of the LFO 1 Amount parameter. A positive value will increase the Amount if the wheel is pushed forward. Negative values invert this relationship.
Phase	This sets modulation wheel control of the Phase Offset parameter for both Osc 1 and 2. Note that Phase Offset Modulation (Subtraction or Multiplication) must be activated for this to have any effect.
FM	This sets modulation wheel control of the FM Amount parameter. A positive value will increase the FM amount if the wheel is pushed forward. Negative values invert this relationship. Both oscillators must be activated for this to have any effect.

Legato

Legato works best with monophonic sounds. Set Polyphony (see below) to 1 and try the following:

- **Hold down a key and press another key without releasing the previous.**
 Notice that the pitch changes, but the envelopes do not start over. That is, there will be no new "attack".
- **If polyphony is set to more voices than 1, Legato will only be applied when all the assigned voices are "used up".**
 For example, if you had a polyphony setting of "4" and you held down a 4 note chord, the next note you played would be Legato. Note, however, that this Legato voice will "steal" one of the voices in the 4 note chord, since all the assigned voices were already used up!

Retrig

This is the "normal" setting for playing polyphonic patches. That is, when you press a key without releasing the previous, the envelopes are retriggered, like when you release all keys and then press a new one. In monophonic mode, Retrig has an additional function; if you press a key, hold it, press a new key and then release that, the first note is also retriggered.

Portamento (Time)

Portamento is when the pitch "glides" between the notes you play, instead of instantly changing the pitch. The Portamento knob is used to set how long it takes for the pitch to glide from one pitch to the next. If you don't want any Portamento at all, set this knob to zero.

Setting Number of Voices - Polyphony



This determines the polyphony, i.e. the number of voices a Subtractor Patch can play simultaneously. This can be used to make a patch monophonic (=a setting of "1"), or to extend the number of voices available for a patch. The maximum number of voices you can set a Subtractor Patch to use is 99. In the (unlikely) event you should need more voices, you can always create another Subtractor!

! **Note that the Polyphony setting does not "hog" voices. For example, if you have a patch that has a polyphony setting of ten voices, but the part the patch plays only uses four voices, this won't mean that you are "wasting" six voices. In other words, the polyphony setting is not something you need to consider much if you want to conserve CPU power - it is the number of voices actually used that counts.**

About the Low Bandwidth button

This can be used to conserve CPU power. When activated, this function will remove some high frequency content from the sound of this particular device, but often this is not noticeable (this is especially true for bass sounds).

External Modulation



Subtractor can receive common MIDI controller messages, and route these to various parameters. The following MIDI messages can be received:

- **Aftertouch (Channel Pressure)**
- **Expression Pedal**
- **Breath Control**

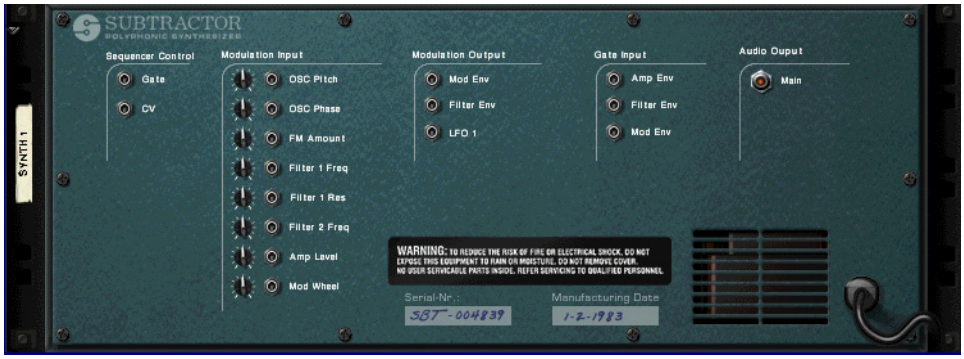
If your MIDI keyboard is capable of sending Aftertouch messages, or if you have access to an Expression Pedal or a Breath controller, you can use these to modulate parameters. The "Ext. Mod" selector switch sets which of these message-types should be received.

These messages can then be assigned to control the following parameters:

Destination	Description
F. Freq	This sets External modulation control of the Filter 1 Frequency parameter. A positive value will increase the frequency with higher external modulation values. Negative values invert this relationship.
LFO 1	This sets External modulation control of the LFO 1 Amount parameter. A positive value will increase the LFO 1 amount with higher external modulation values. Negative values invert this relationship.

Destination	Description
Amp	This let's you control the overall volume of the sound with external modulation. If a positive value is set, the volume will increase with higher external modulation values. A negative value inverts this relationship.
FM	This sets External modulation control of the FM Amount parameter. If a positive value is set, the FM amount will increase with higher external modulation values. A negative value inverts this relationship. Both oscillators must be activated for this to have any effect.

Connections



Flipping the Subtractor around reveals a plethora of connection possibilities, most of which are CV/Gate related. Using CV/Gate is described in the chapter "Routing Audio and CV".

Audio Output

This is Subtractor's main audio output. When you create a new Subtractor device, this is auto-routed to the first available channel on the audio mixer.

Sequencer Control

The Sequencer Control CV and Gate inputs allow you to play the Subtractor from another CV/Gate device (typically a Matrix or a Redrum). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity.

! **For best results, you should use the Sequencer Control inputs with monophonic sounds.**

Modulation Inputs

! **Remember that CV connections will not be stored in the Subtractor patch, even if the connections are to/from the same Subtractor device!**

These control voltage (CV) inputs (with associated voltage trim pots), can modulate various Subtractor parameters from other devices, or from the modulation outputs of the same Subtractor device. These inputs can control the following parameters:

- **Oscillator Pitch (both Osc 1 & 2).**
- **Oscillator Phase Offset (both Osc 1 & 2).**
- **FM Amount**
- **Filter 1 Cutoff**
- **Filter 1 Res**
- **Filter 2 Cutoff**
- **Amp Level**
- **Mod Wheel**

Modulation Outputs

The Modulation outputs can be used to voltage control other devices, or other parameters in the same Subtractor device. The Modulation Outputs are:

- **Mod Envelope**
- **Filter Envelope**
- **LFO 1**

Gate Inputs

These inputs can receive a CV signal to trigger the following envelopes. Note that connecting to these inputs will override the normal triggering of the envelopes. For example, if you connected an LFO output to the Gate Amp input, you would not trigger the amp envelope by playing notes, as this is now controlled by the LFO. In addition you would only hear the LFO triggering the envelope for the notes that you hold down. The following Gate Inputs can be selected:

- **Amp Envelope**
- **Filter Envelope**
- **Mod Envelope**

Chapter 18
Malström Synthesizer

Introduction

The Malström is a polyphonic synthesizer with a great number of different routing possibilities. It is based on the concept of what we call "Graintable Synthesis" (see below), and is ideally suited for producing swirling, sharp, distorted, abstract special effect types of synthesizer sounds. In fact, you could go so far as to say that the Malström can produce sounds quite unlike anything you've ever heard from a synthesizer.

For a complete run-down of the principles behind it and thorough explanations of the controls, read on...

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Features

The following are the basic features of the Malström:

- **Two Oscillators, based on Graintable Synthesis.**
 See "The Oscillator section" for details.
- **Two Modulators, featuring tempo sync and one-shot options.**
 See "The Modulator section".
- **Two Filters and one Shaper.**
 A number of different filter modes in combination with several routing options and a Waveshaper makes it possible to create truly astounding filter effects.
- **Three Envelope generators.**
 There is one amplitude envelope for each oscillator and a common envelope for both filters. See "The amplitude envelopes" and "The Filter Envelope" for details.
- **Polyphony of up to 16 voices.**
- **Velocity and Modulation control.**
 See "The Velocity controls".
- **A number of CV/Gate Modulation possibilities.**
 See "Modulation Input".
- **A variety of Audio Input/Output options.**
 You can for instance connect external audio sources for input to the Malström, and you can also control its output. See "Audio Input" for more details.

Theory of operation

There are a number of different synthesis methods for generating sound, e.g. subtractive synthesis (which is used in the Subtractor), FM synthesis, and physical modelling synthesis to mention but a few.

To give you a clear understanding of the inner workings of the Malström, it might be in order with a brief explanation of what we call Graintable Synthesis.

What we refer to as graintable synthesis is actually a combination of two synthesis methods, granular synthesis and wavetable synthesis.

- **In granular synthesis, sound is generated by a number of short, contiguous segments (grains) of sound, each typically from 5 to 100 milliseconds long.**
 The sound is varied by changing the properties of each grain and/or the order in which they are spliced together. Grains can be produced either by a mathematical formula or by a sampled sound. This is a very dynamic synthesis method capable of producing a great variety of results, although somewhat hard to master and control.
- **Wavetable synthesis on the other hand, is basically the playback of a sampled waveform.**
 An oscillator in a wavetable synth plays back a single period of a waveform, and some wavetable synths also allow for sweeping through a set of periodic waveforms. This is a very straightforward synthesis method, easily controlled, but somewhat more restricted in results. The Malström combines these two into a synthesis method that provides a very flexible way of synthesizing sounds with incredible flux and mutability.

The Malström combines these two into a synthesis method that provides a very flexible way of synthesizing sounds with incredible flux and mutability.

It works like this:

- **The oscillators in the Malström play back sampled sounds that have been subjected to some very complex processing and cut up into a number of grains.**
- **A set of these periodic waveforms (grains) are spliced together to form a Graintable, which may be played back to reproduce the original sampled sound.**
- **A Graintable may be treated just like a wavetable; e.g., you may choose to sweep through it, to move through it at any speed without affecting pitch, to play any section of it repeatedly, to select from it static waveforms, to jump between positions, etc., etc.**
- **It is also possible to perform a number of other tricks, all of which are described further on in this chapter.**

Loading and Saving Patches

Loading and saving patches is done in the same way as with any other Reason device - see “Loading patches” and “Saving patches”.

The Oscillator section



The two oscillators (osc:A and osc:B) of the Malström are the actual sound generators, and the rest of the controls are used for modulating and shaping the sound. The oscillators actually do two things; they play a graintable and generate the pitch:

- **A graintable is several short, contiguous segments of audio (see above).**
- **Pitch is the frequency at which the segments are played back.**

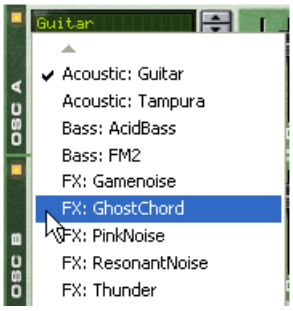
When creating a Malström patch, the fundamental first building block is usually to select a graintable for one or both of the oscillators.

→ **To activate/deactivate an oscillator, click the On/Off button in the top left corner.**
When an oscillator is activated, the button is lit.



An activated oscillator.

→ **To select a graintable, either use the spin controls or click directly in the display to bring up a pop-up menu with the available graintables.**
The graintables are sorted alphabetically into a number of descriptive categories, giving a hint as to the general character of the sound. Note that the categories are only visible in the pop-up menu, not in the display.



Selecting a graintable by clicking in the Oscillator display.

Setting oscillator frequency

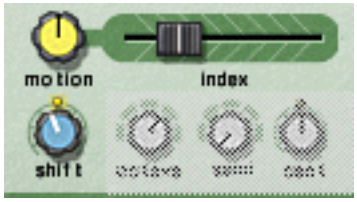
You can change the frequency - i.e. the tuning - of each oscillator by using the three knobs marked “Octave”, “Semi” and “Cent”.



- **The Octave knob changes the frequency in steps of one full octave (12 semitones).**
 The range is -4 – 0 – +4 where 0 corresponds to middle “A” on your keyboard at 440 Hz.
- **The Semi knob changes the frequency in steps of one semitone.**
 The range is 0 to +12 (one full octave up).
- **The Cent knob changes the frequency in steps of cents, which are 100ths of a semitone.**
 The range is -50 – 0 – +50, i.e. down or up by up to half a semitone.

Controlling playback of the graintable

Each oscillator features three controls that determine how the loaded graintables are played back. These are: The “Index” slider, the “Motion” knob and the “Shift” knob.



- **The Index slider sets the playback starting point in the graintable.**
 By dragging the slider, you set which index point in the graintable should be played first when the Malström receives a Note On message. Playback will then continue to the next index point according to the active graintable. With the slider all the way to the left, the first segment in the graintable is also the one that will be played back first.

! **Note that the Malström’s Graintables are *not* all of the same length, and that the range for the Index slider (0-127) does not reflect the actual length of the graintables. I.e. regardless of whether a graintable contains 3 or 333 grains, the Index slider will always span the entire graintable even though the slider range says 0-127.**

- **The Motion knob controls how fast the Malström should move forward to play the next segment in the graintable, according to its motion pattern (see below).**
 If the knob is kept in the middle position the speed of motion is the normal default. Turning the knob to the left slows it down and turning it to the right results in higher speed. If the knob is set all the way to the left, there will be no motion at all, which means that the initial segment, as set with the Index slider, will play over and over as a static waveform.
- **The Shift knob changes the timbre of the sound (the formant spectrum).**
 What it actually does is change the pitch of a segment up or down by re-sampling. However, since the pitch you *hear* is independent of the actual pitch of the graintable (see above), pitch-shifting a segment instead means that more or less of the segment waveform will be played back, resulting in a change of harmonic content and timbre.

About motion patterns

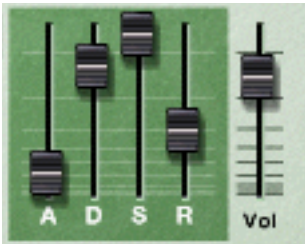
Each graintable has a predefined motion pattern and a default motion speed.

When a graintable is looped (i.e. if the Motion knob is *not* set all the way to the left), it follows one of two possible motion patterns:

- **Forward**
 This motion pattern plays the graintable from the beginning to the end, and then repeats it.
- **Forward - Backward**
 This motion pattern plays the graintable from the beginning to the end, then from the end to the beginning and then repeats it.

The motion speed can be changed with the Motion knob, as described above, but it is *not* possible to alter the motion pattern of a graintable.

The amplitude envelopes



Each oscillator features a standard ADSR (Attack, Decay, Sustain, Release) envelope generator, and a Level control. These are used for controlling the volume of the oscillator. One thing that makes the Malström different from many other synths though, is the fact that the amplitude envelopes are placed *before* the filter and routing sections in the signal path.

The amplitude envelopes control how the volume of a sound should change from the moment you strike a key on your keyboard to the moment that you release it again.

Vol

The Volume knobs set the volume level out from each oscillator.

! **For an overall description of the general envelope parameters (Attack, Decay, Sustain, Release), please refer to the Subtractor chapter.**

The Modulator section



The Malström features two Modulators (mod:A and mod:B) These are in fact another type of oscillators, called LFOs (Low Frequency Oscillators). They each generate a waveform and a frequency, much like osc:A and osc:B. However, there are a couple of important differences:

- **Mod:A and mod:B do not generate sound. They are instead used for modulating various parameters to change the character of the sound.**
- **They only generate waveforms of low frequency.**

Furthermore, both modulators are tempo syncable and possible to use in one shot mode, in which case they will actually work like envelopes.

Modulator parameters

The two Modulators have a few controls in common, but there are also some differences. Both the common parameters and the ones that are unique for each Modulator (the destinations) are described below.

→ **To activate/deactivate a Modulator, click the On/Off button in the top left corner.**
When a Modulator is activated, the button is lit.



An activated Modulator.

Curve

This lets you select a waveform for modulating parameters. Use the spin controls to the right of the display to cycle through the available waveforms. Some of these waveforms are especially suited for use with the Modulator in one shot mode (see below).

Rate

This knob controls the frequency of the Modulator. For a faster modulation rate, turn the knob to the right.

The Rate knob is also used for setting the time division when synchronizing the Modulator to the song tempo (see below).

One Shot

To put the Modulator into one shot mode, click this button so that it is lit.

Normally, the Modulators will repeat the selected waveforms over and over again, at the set rate. However, when one shot mode is activated and you play a note, the Modulator will play the selected waveform only once (at the set rate) and then stop. In other words, it will effectively be turned into an envelope generator!

Note that even though all waveforms can be used with interesting results, some waveforms are explicitly well suited for use in one shot mode. For example, try using the waveform with just one long, gently sloping curve.

Sync

Clicking this button so that it is lit synchronizes the Modulator to the song tempo, in one of 16 possible time divisions.

! **When sync is activated, the Rate knob is used for selecting the desired time division. Turn the Rate knob and observe the tool tip for an indication of the time division.**

A/B selector



This switch is used for deciding which oscillator and/or filter the Modulator should modulate - A, B or both. With the switch in the middle position, both A and B will be modulated.

Destinations

The following knobs are used for determining what each of the two modulators should modulate.

- **Note that these knobs are bi-polar, which means that if a knob is in the middle position, no modulation is applied. If you turn a knob either to the left or to the right, an increasing amount of modulation is applied to the parameter. The difference is that if you turn a knob to the left, the waveform of the modulator is inverted.**

Mod:A



Mod:A can modulate the following parameters of either oscillator:

- **Pitch**
 Use this if you want Mod:A to offset the pitch of osc:A, osc:B, or both (see "Setting oscillator frequency").
- **Index**
 Use this if you want Mod:A to offset the index start position of osc:A, osc:B, or both (see "Controlling playback of the graintable").
- **Shift**
 Use this to have Mod:A affect the harmonic content of osc:A, osc:B, or both (see "Controlling playback of the graintable").

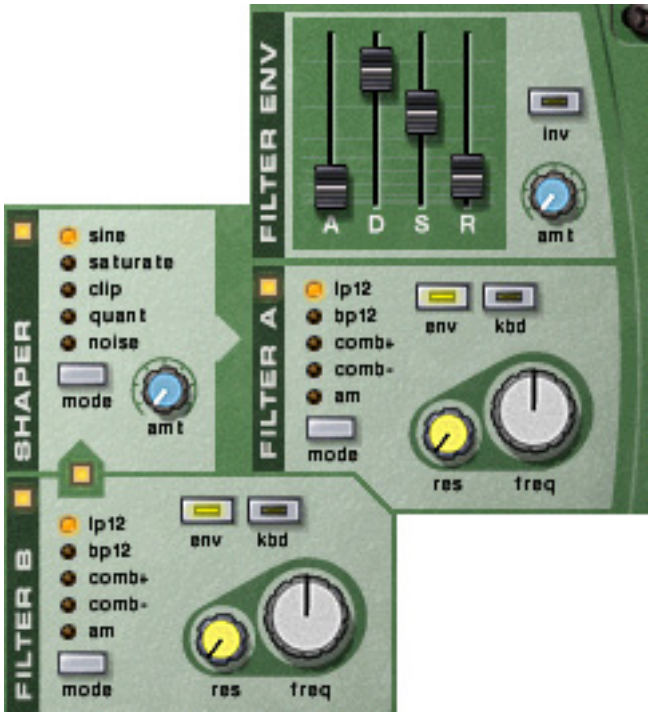
Mod:B



Mod:B can modulate the following parameters of either oscillator:

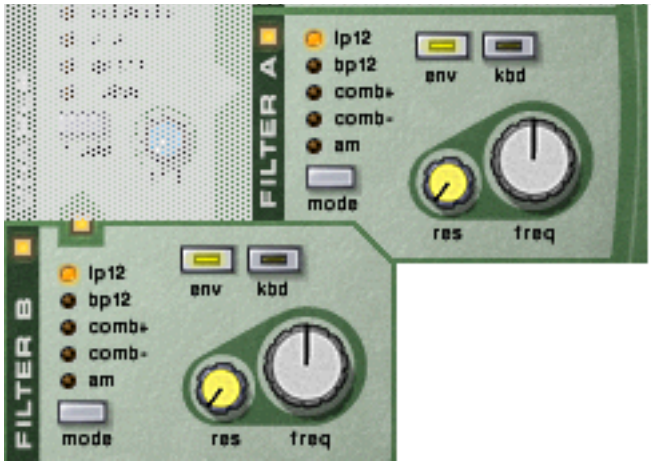
- **Motion**
 Use this if you want Mod:B to affect the motion speed of osc:A, osc:B, or both (see "Controlling playback of the graintable").
- **Vol**
 Use this if you want Mod:B to change the output level of osc:A, osc:B, or both (see "Vol").
- **Filter**
 Use this if you want Mod:B to offset the cutoff frequency of filter:A, filter:B, or both (see "Filter controls").
- **Mod:A**
 Use this if you want Mod:B to change the total amount of modulation from Mod:A.

The Filter section



The filter section lets you further shape the overall character of the sound. Contained herein are two multimode filters, a filter envelope and a waveshaper.

The Filters



Both filter:A and filter:B have the exact same parameters, all of which are described below.

→ **To activate/deactivate a filter, click the On/Off button in the top left corner.**
When a filter is activated, the button is lit.



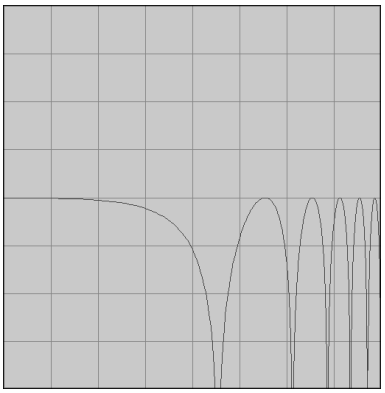
An activated filter.

Filter types

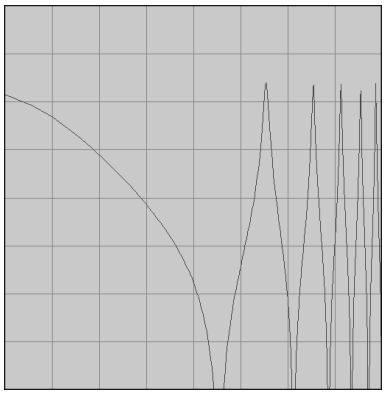
To select a filter type, either click the Mode button in the bottom left corner or click directly on the desired filter name so that it lights up in yellow:

- **LP 12 (12 dB lowpass)**
 Lowpass filters let low frequencies through and cut off high frequencies. This filter type has a roll-off curve of 12dB/Octave.
- **BP 12 (12 dB bandpass)**
 Bandpass filters cut both high and low frequencies, leaving the frequency band in between unaffected. Each slope in this filter type has a 12 dB/Octave roll-off.

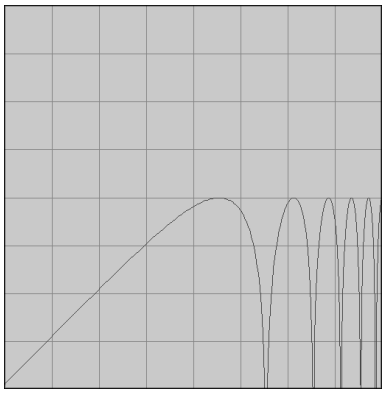
- **Comb + & Comb –**
 Comb filters are basically delays with very short delay times with adjustable feedback (in Reason controlled with the Resonance knob). A comb filter causes resonating peaks at certain frequencies.
 The difference between "+" and "–" is in the position of the peaks, in the spectrum. The main audible difference is that the "–"-version causes a bass cut.
 The Resonance parameter in both cases controls the shape and size of the peaks.



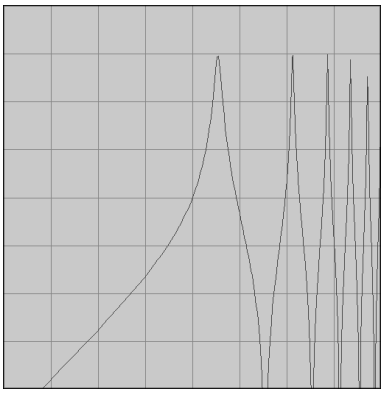
Comb + Low Resonance.



Comb + High Resonance.



Comb – Low Resonance.



Comb – High resonance.

- **AM**
 AM (Amplitude Modulation) is often referred to as Ring Modulation. A Ring Modulator works by multiplying two signals together. In the case of Malström, the filter produces a sine wave which is multiplied with the signal from osc:A or osc:B. Resonance controls the mix between the clean and modulated signals. The Ring Modulated output will then contain added frequencies which are generated by the sum of, and the difference between the two signals. This can be used for creating complex, non-harmonic sounds.

Filter controls

Each filter contains the following four controls:

- **Kbd (keyboard tracking)**
 By clicking this button so that it is lit, you activate keyboard tracking. If keyboard tracking is activated, the frequency of the filter will change according to the notes you play on your keyboard. That is, if you play notes higher up on the keyboard, the filter frequency will increase and vice versa. If keyboard tracking is deactivated, the filter frequency will remain at a fixed value regardless of where on the keyboard you play.

- **Env (envelope)**
 If you click on this button so that it is lit, the cutoff frequency (see below) will be modulated by the filter envelope. If you leave this deactivated, the Filter Envelope will have no effect.
- **Freq (frequency)**
 The function of this parameter depends on which filter type you have selected:

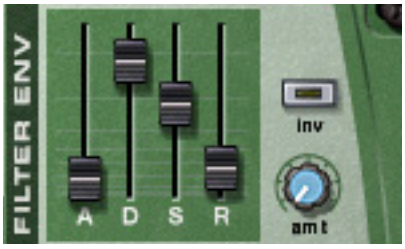
 With all filter types except AM, it is used for setting the cutoff frequency of the filter. In the case of the lowpass filter for example, the cutoff frequency determines the limit above which high frequencies will be cut off. Frequencies below the cutoff frequency will be allowed to pass through. The farther to the right you turn the knob, the higher the cutoff frequency will be.

 If you have selected AM as filter type, this will instead control the frequency of the signal generated by the filter. The same control range applies though; the farther to the right you turn the knob the higher the frequency will be.
- **Res (resonance)**
 Again, the function of this parameter depends upon which filter type is selected:

 If the selected filter is any other type than AM, it sets the filter characteristic, or quality. For the lowpass filter for example, raising the filter Res value will emphasize the frequencies around the set filter frequency. This generally produces a thinner sound, but with a sharper, more pronounced filter frequency "sweep". The higher the filter Res value, the more resonant the sound becomes until it produces a whistling or ringing sound. If you set a high value for the Res parameter and then vary the filter frequency, this will produce a very distinct sweep, with the ringing sound being very evident at certain frequencies.

 In the case of the AM filter type though, this control instead regulates the balance between the original signal and the signal resulting from amplitude modulation. The farther to the right you turn the knob, the more dominant the AM signal will be.

The Filter Envelope



This is a standard ADSR envelope with two additional controls; inv and amt. The filter envelope is common for both filter:A and filter:B, and controls how the filter frequency should change over time.

Inv (inverse)

This button toggles inversion of the envelope on and off. The Decay segment of the envelope will for instance normally lower the frequency, but if the envelope is inverted it will instead raise the frequency.

Amt (amount)

This controls to which extent the filter envelope affects the filters, or rather - the set filter cutoff frequencies. For example; if the cutoff frequency is set to a certain value, the filter will already be opened by this amount when you hit a key on your keyboard. The amount setting then controls how much more the filter will open from that point. Turn the knob to the right to increase the value.

! **For an overall description of the general envelope parameters (Attack, Decay, Sustain, Release), please refer to the Subtractor chapter.**

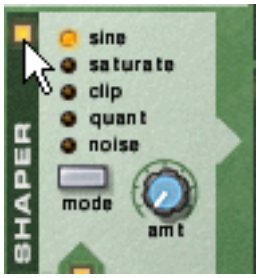
The Shaper



Before filter:A is an optional waveshaper. Waveshaping is a synthesis method for transforming sounds by altering the waveform shape, thereby creating a complex, rich sound. Or, if that's more to your taste, truncating and distorting the sound to lo-fi heaven!

A guitar distortion box could be viewed as a type of waveshaper for example. An unamplified electric guitar produces a sound with fairly pure harmonic content, which is then amplified and transformed by the distortion box.

→ **To activate/deactivate the Shaper, click the On/Off button in the top left corner.**
When the Shaper is activated, the button is lit.



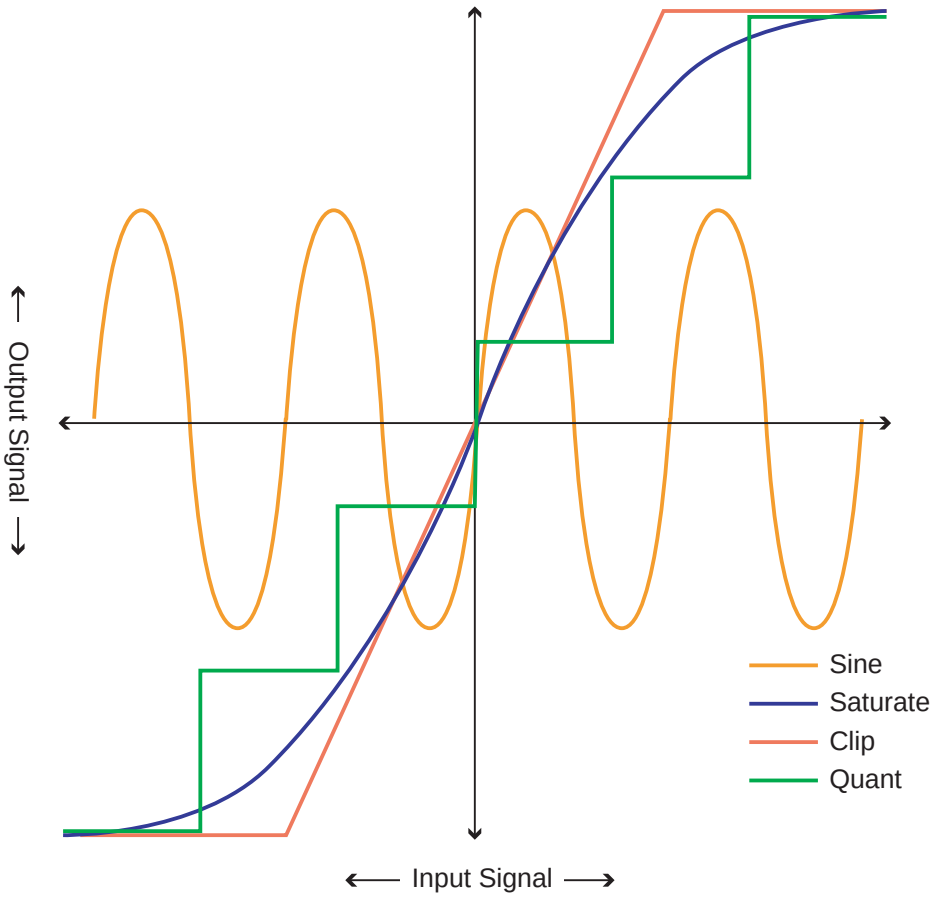
The Shaper activated

Mode

You can select one of five different modes for shaping the sound, each with its own characteristics.

To select a mode, either click the Mode button in the bottom left corner or click directly on the desired mode name so that it lights up in yellow.

- **Sine**
 This produces a round, smooth sound.
- **Saturate**
 This gives a lush, rich character to the sound.
- **Clip**
 This introduces clipping - digital distortion - to the signal.
- **Quant**
 This lets you truncate the signal by bit-reduction, thus making it possible to achieve that noisy, characteristic 8 bit sound for example.
- **Noise**
 This is actually not strictly a shaper function. Instead it multiplies the sound with noise.



Amt (amount)

This controls the amount of shaping applied. By turning the knob to the right you increase the effect.

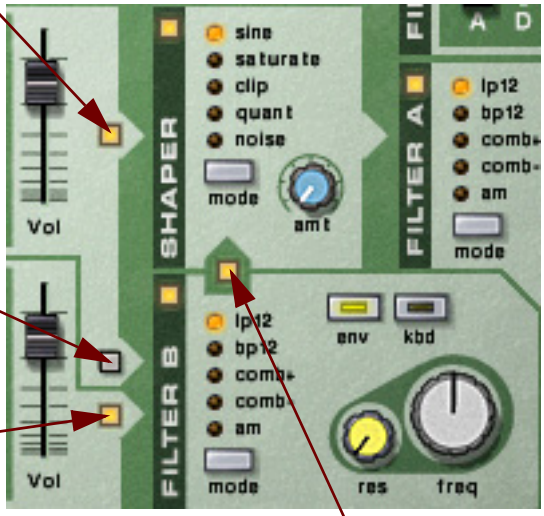
Routing

The Malström puts you in total control of how the signal should be routed from the oscillators, through the filters and on to the outputs. Below is first a general description of the routing options, followed by examples of how to route the signal in order to achieve a certain result.

→ **Click on a button so that it is lit, to route the signal correspondingly.**
See below for descriptions.

If this button is lit, the signal from osc:A is routed to filter:A via the shaper. If neither this nor the other routing button from osc:A (to filter:B) is lit, the signal will go straight to the outputs.

If this button is lit, the signal from osc:A is routed to filter:B. If neither this nor the other routing button from osc:A (to filter:A/shaper) is lit, the signal from osc:A will go straight to the outputs.



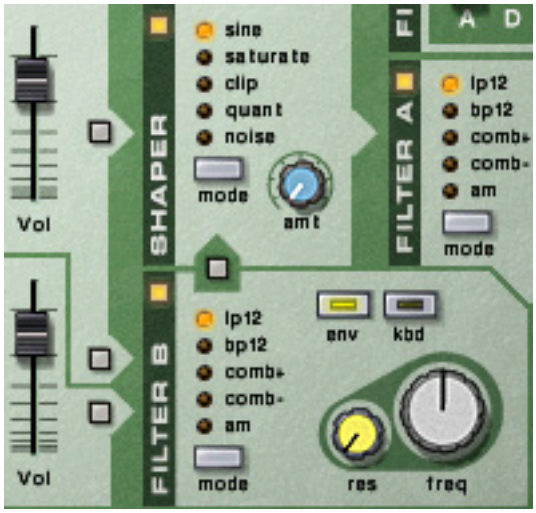
If this button is lit, the signal from osc:B is routed to filter:B. If this is not lit, the signal from osc:B will go straight to the outputs.

If this button is lit, the signal from filter:B is routed to filter:A via the shaper. The signal from filter:B can originate from either osc:A, osc:B or both. If this is not lit, the signal from filter:B will go straight to the outputs.

! **Note that the result depends both on the routing buttons and on whether the filters and shaper are activated or not!**

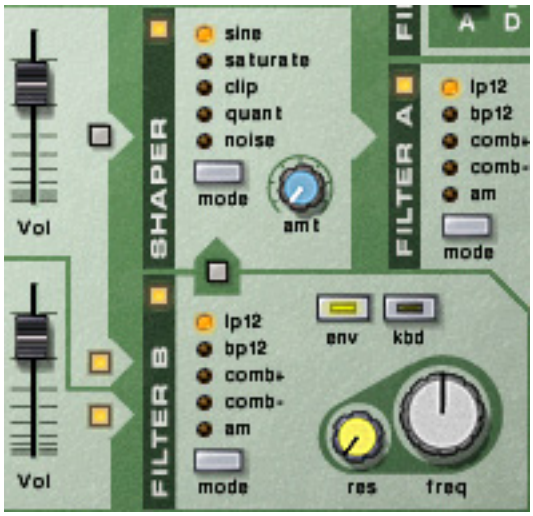
Routing examples

One or both oscillators without filters

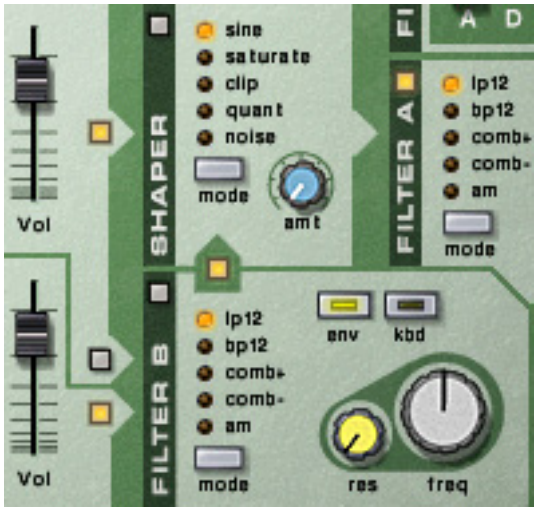


With this configuration, the signals from the oscillators will bypass the filters and the shaper and go directly to the respective output. Using both oscillators allows you to use the Spread parameter to create a true stereo sound.

One or both oscillators to one filter only



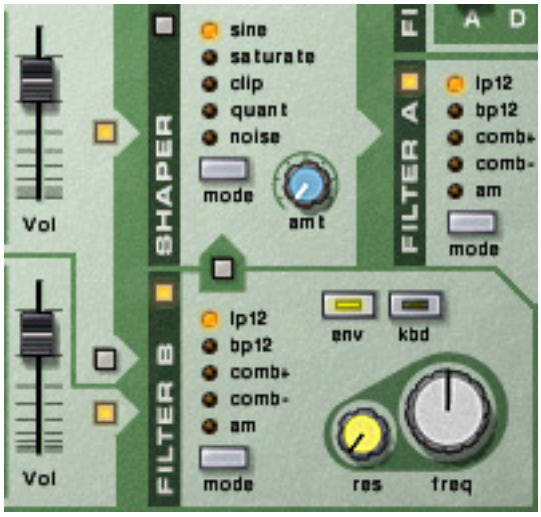
Both oscillators routed to filter:B only.



Both oscillators routed to filter:A only.

With these configurations, the signal from osc:A and/or osc:B will go to either filter:A or filter:B and then to the outputs. This is essentially a mono configuration and hence Spread should probably be set to “0”.

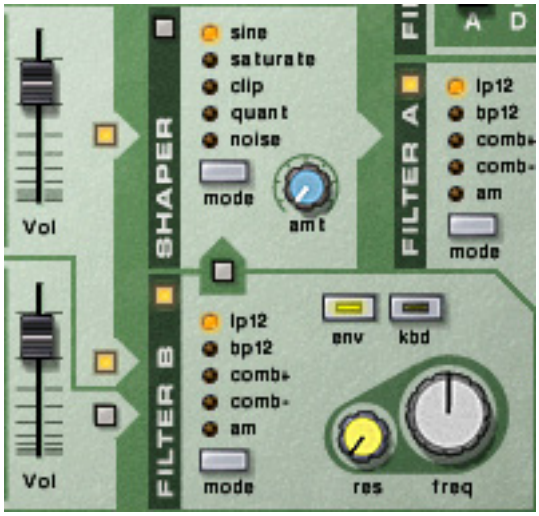
Both oscillators to one filter each



With this configuration, the signals from osc:A and osc:B will go to filter:A and filter:B respectively, and then to the outputs.

Again, this configuration allows you to work in true stereo.

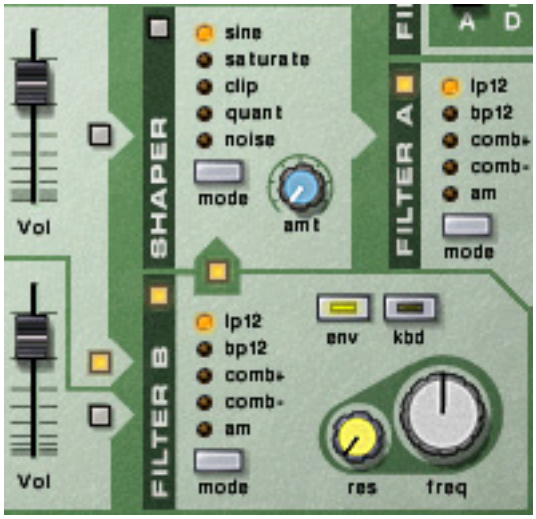
Oscillator A to both filters in parallel



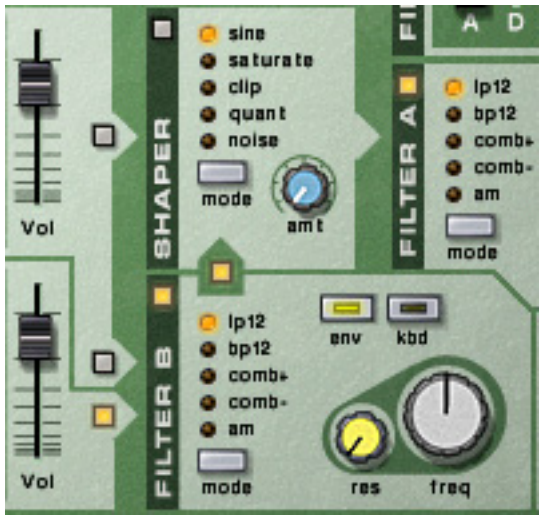
With this configuration, the signal from osc:A will go to both filter:A and filter:B, with the filters in parallel.

! **This configuration is only possible with osc:A. Osc:B can be routed to both filters as well, but only in series (see below).**

One or both oscillators with both filters in series



Osc:A routed through both filters in series.

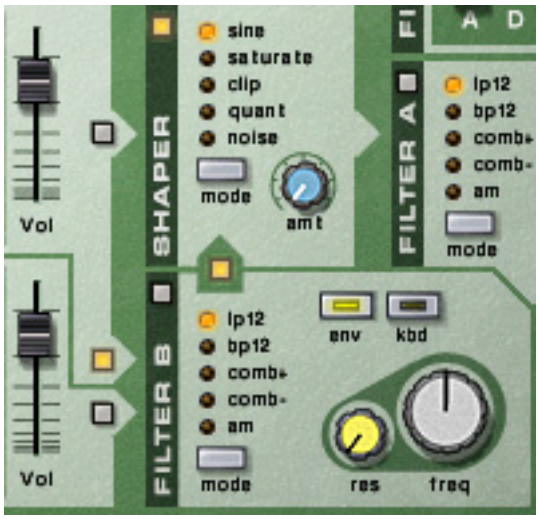


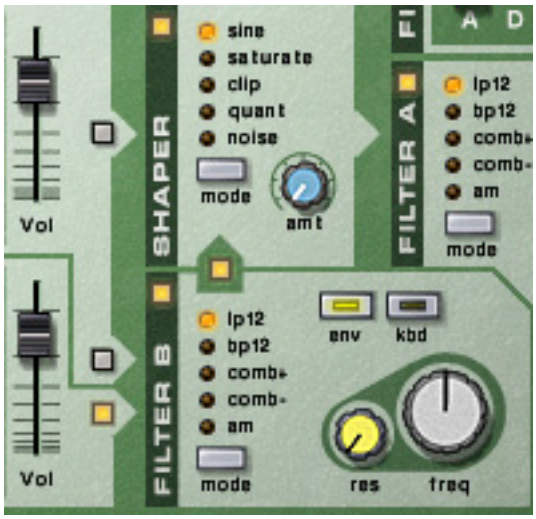
Osc:B routed through both filters in series.

With these configurations, the signal from osc:A and/or osc:B will go to both filter:A and filter:B, with the filters in series (one after the other).

Adding the shaper

The signal from one or both oscillators can also be routed to the shaper. The signal will then pass through the shaper to the outputs, with or without also passing through the filters.





In the left figure, the signal from osc:A is routed to the shaper and then directly to the outputs. In the right figure, the signal from osc:B is routed to filter:B, then to the shaper and then to filter:A.

The output controls



These two parameters control the output from the Malström in the following way:

Volume

This knob controls the master volume out from the Malström.

Spread

This controls the stereo pan-width of the outputs from Osc:A/B and Filter:A/B respectively. The farther to the right you turn the knob, the wider the stereo image will be. In other words, the signals will be panned further apart to the left and right.

! **If you are only using one output (A or B), it is strongly recommended that you set Spread to "0".**

The play controls



To the far left on the Malström's "control panel" are various parameters that are affected by how you play, and lets you apply modulation by MIDI controls. The following is a description of these controls.

Polyphony - setting the number of voices



This lets you set the polyphony for the Malström. Polyphony is the number of voices it can play simultaneously. The maximum number is 16 and the minimum is 1, in which case the Malström will be monophonic.

! **The number of voices you can play depends of course on the capacity of your computer. Even though the maximum number is 16 it doesn't necessarily mean that your system is capable of using that many voices. Also note that voices do not consume CPU capacity unless they are really "used". That is, if you are using a patch that plays two voices but have polyphony set to four, the two "unused" voices do not consume any of your system resources.**

Porta (portamento)



This is used for controlling portamento. This is a parameter that makes the pitch glide between the notes you play, rather than changing the pitch instantly as soon as you hit a key on your keyboard. By turning this knob you set how long it should take for the pitch to glide from one note to the next as you play them.

With the knob turned all the way to the left, portamento is disabled.

Legato



By clicking this button you activate/deactivate Legato. Legato in Malström is unique in that it allows you to control whether the sound is monophonic or polyphonic by using your playing style:

→ **If you play legato (hold down a key and then press another key without releasing the previous), the sound is monophonic.**
Also note that the pitch changes, but the envelopes do not start over. That is, there will be no new "attack".

→ **If you play non-legato (separated notes), with polyphony set to more voices than 1, each note will decay separately (polyphonic).**
This will be most apparent with longer release times.

The Pitch Bend and Modulation wheels



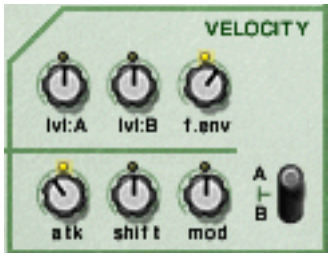
- **The Pitch Bend wheel is used for bending the pitch of notes, much like bending the strings on a guitar or other string instrument.**
- **The Modulation wheel can be used for applying modulation while you are playing.**

Virtually all MIDI keyboards have Pitch Bend and Modulation controls. The Malström does not only feature the settings for how incoming MIDI Pitch Bend and Modulation wheel messages should affect the sound, but also two functional wheels that can be used for applying real time modulation and pitch bend if you don't have these controllers on your keyboard, or if you aren't using a keyboard at all. The wheels on the Malström also mirror the movements of the wheels on your MIDI keyboard.

Pitch Bend Range

The Range parameter sets the maximum amount of pitch bend, i.e. how much it is possible to change the pitch by turning the wheel fully up or down. The maximum range is 24 semitones (2 Octaves). You change the value by clicking the spin controls to the right of the display.

The Velocity controls



Velocity is used for controlling various parameters according to how hard or soft you play notes on your keyboard. A typical use of velocity control is to make sounds brighter and louder if you strike a key harder. By using the knobs in this section, you can control how much the various parameters will be affected by velocity.

! **All of the velocity control knobs are bi-polar, which means that the amount can be set to either positive or negative values, while keeping the knobs in the center position means that no velocity control is applied.**

The following parameters can be velocity controlled:

- **Lvl:A**
 This lets you velocity control the output level of osc:A.
- **Lvl:B**
 This lets you velocity control the output level of osc:B.

- **F.env**
 This sets velocity control for the Filter Envelope Amount parameter. Positive values will increase the envelope amount the harder you play, and negative values will decrease the amount.
- **Atk (attack)**
 This sets velocity control for the Amp Envelope Attack parameter of osc:A and/or osc:B. Positive values will increase the Attack time the harder you play, and negative values will decrease it.
- **Shift**
 This lets you velocity control the Shift parameter of osc:A and/or osc:B.
- **Mod**
 This lets you velocity control all modulation amounts of mod:A and/or mod:B.

! **Note that you can set the last three parameters (Atk, Shift and Mod) to be velocity controlled for either or both of oscillator/modulator A and B. This is done with the A/B selector switch.**

The Modulation wheel controls



The Modulation wheel can be set to control a number of parameters. You can set positive or negative values, just like in the Velocity Control section (see above).

The following parameters can be affected by the modulation wheel:

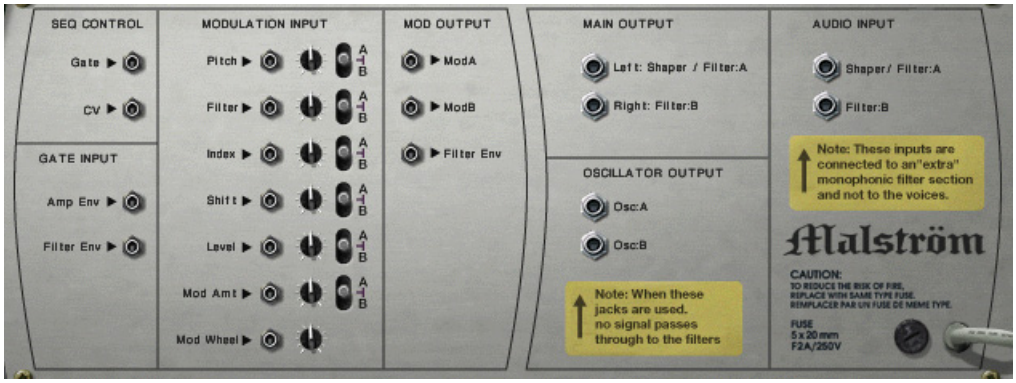
- **Index**
 This sets modulation wheel control of the currently active graintable's index (see "Controlling playback of the graintable") for osc:A and/or osc:B. Positive values will move the index position forwards if the modulation wheel is pushed forward. Negative values will move it backwards.
- **Shift**
 This sets modulation wheel control of the Shift parameter of osc:A and/or osc:B (see "Controlling playback of the graintable").
- **Filter**
 This sets modulation wheel control of the Filter Frequency

 parameter (see "Filter controls"). Positive values will raise the frequency if the

 wheel is pushed forward and negative values will lower the frequency.
- **Mod**
 This sets modulation wheel control of the total amount of modulation from mod:A and/or mod:B. Positive values will increase the settings if the wheel is pushed forward and negative values will decrease the settings.

! **You can set whether these parameters on either or both oscillator/modulator/filter A and B will be affected by the modulation wheel. This is done with the A/B selector switch.**

Connections



Flipping the Malström around reveals a wide array of connection possibilities. Most of these are CV/Gate related. Using CV/Gate is described in the chapter "Routing Audio and CV".

Audio Output

These are the Malström's audio outputs. When you create a new Malström device, they are auto-routed to the first available channel on the audio mixer:

- **Shaper/Filter:A (left) & Filter:B (right)**
 These are the main stereo outputs. Each of the two filters are connected to a separate output, and by connecting both, you can have stereo output. Whether the output really will be in stereo however, is determined by the routing and the Spread parameter. See "Routing" for details about this.
- **Osc:A & osc:B**
 These make it possible to output the sound directly after the Amp Envelope of each oscillator, bypassing the filter section. Connecting one or both of these to a channel on the audio mixer will break the Malström's internal signal chain. That is, it is not possible to process the sound by using the filters and the shaper of the Malström. the sound instead goes directly to the mixer.

► **Note also that you can connect the outputs Osc:A & Osc:B to the Audio Inputs on the Malström for some interesting effects - see "Routing external audio to the filters".**

Audio Input

- **Shaper/Filter:A**
- **Filter:B**

These inputs let you connect either other audio sources, or the Malström's own internal signal directly to the filters and the shaper - see "Routing external audio to the filters".

Sequencer Control

The Sequencer Control CV and Gate inputs allow you to play the Malström from another CV/Gate device (typically a Matrix or a Redrum). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity.

! **For best results, you should use the Sequencer Control inputs with monophonic sounds.**

Gate Input

These inputs can receive a CV signal to trigger the following envelopes:

- **Amp Envelope**
- **Filter Envelope**

! **Note that connecting to these inputs will override the normal triggering of the envelopes. For example, if you connected a Modulation output to the Gate Amp in-put, you would not trigger the amp envelope by playing notes, as this is now controlled by the Modulator. In addition you would only hear the Modulator triggering the envelope for the notes that you hold down.**

Modulation Input

These control voltage (CV) inputs (with associated voltage trim pots and A/B selector switches), can modulate various Malström parameters from other devices, or from the modulation outputs of the same Malström device. These inputs can control the following parameters:

- **Oscillator Pitch**
- **Filter Frequency**
- **Oscillator Index offset**
- **Oscillator Shift**
- **Amp Level**
- **Mod Amount**
- **Mod Wheel**

Modulation Output

The Modulation outputs can be used to voltage control other devices, or other parameters in the same Malström device.

The Modulation Outputs are:

- **Mod:A**
- **Mod:B**
- **Filter Envelope**

Routing external audio to the filters



The audio inputs on the back of the Malström allows you to connect any audio signal to the filters and Shaper.

To use this feature, it's important to understand the following background:

Normally the Malström behaves like any regular polyphonic synthesizer, in that each voice has its own filter. The filter settings are the same, but each filter envelope is triggered individually when you play a note.

However, when you connect a signal to the audio inputs, it is routed to an "extra" filter. The envelope for this filter is triggered each time any of the other filter envelopes is triggered. In other words, the "extra" filter envelope is triggered each time you play a note on the Malström.

There are two different uses for the audio inputs:

Connecting an external signal source

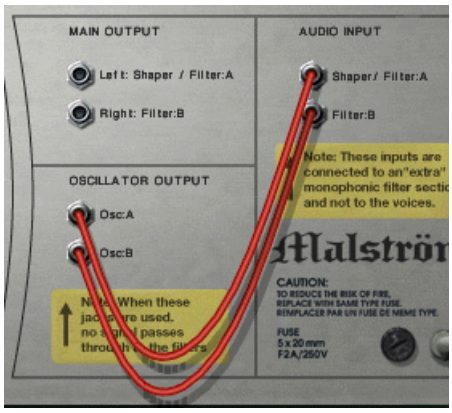
Connecting an audio signal from another device in the rack to the audio input allows you to process the signal through the filters and/or Shaper of the Malström. The processed signal will then be mixed with the Malström's "own" voices (if activated) and sent to the outputs.

The result depends on the following:

- **To which jack you connect the signal.**
- **Whether the filters and/or Shaper are activated on the front panel.**
- **The routing button for filter:B.**
 If this is activated and you connect a signal to the Filter:B input, the signal will be processed in filter:B and then sent to the Shaper and filter:A (just as when routing Malström's own oscillators on the front panel).

Note again that the filter envelope is triggered by all voices. To make use of the filter envelope, you either need to play the Malström or use gate signals to trigger it or the filter envelope, separately.

Connecting the signals from the Malström itself



If you connect one or both oscillator outputs to the audio input(s), the internal signal path from the oscillators to the filters is broken. In other words, no signals will pass internally from the oscillators to the filters, and the three routing buttons for the oscillators are ignored.

This may seem pointless at first, but there are several uses for this:

- **When you play the Malström in this mode, the filter envelope will be triggered for each note you play, affecting all sounding notes.**
 This is due to the monophonic "extra" filter described above. On older synthesizers, this feature is called "Multiple triggering".
- **Since all notes you play are mixed before being sent into the filter, the result of using the Shaper will be totally different (if you play more than one note at a time).**
 This is similar to playing a guitar chord through a distortion effect, for example.
- **You can patch in external effects between the oscillators and the filters.**
 Just connect an oscillator output to the input of the effect device, and the effect output to the Malströms's audio input.
- **You can use combinations of connections and routing. You could for instance connect an external audio signal to one of the inputs, one of the Malström's oscillators to the other input and then use the routing options on the front panel for the other oscillator. All of these signals will then be mixed and sent to the Malström's main outputs.**

Chapter 19
Monotone
Bass Synthesizer

Introduction



The Monotone Bass Synthesizer is a great little monophonic bass synthesizer. Despite its fairly small size, it's very versatile can produce really fat and punchy bass synth sounds.

Monotone features two oscillators, a 24 dB lowpass ladder filter, amp envelope and chorus and delay effects. It also has an LFO and an additional envelope for modulation purposes.

Panel overview

The Monotone front panel contains the following sections:

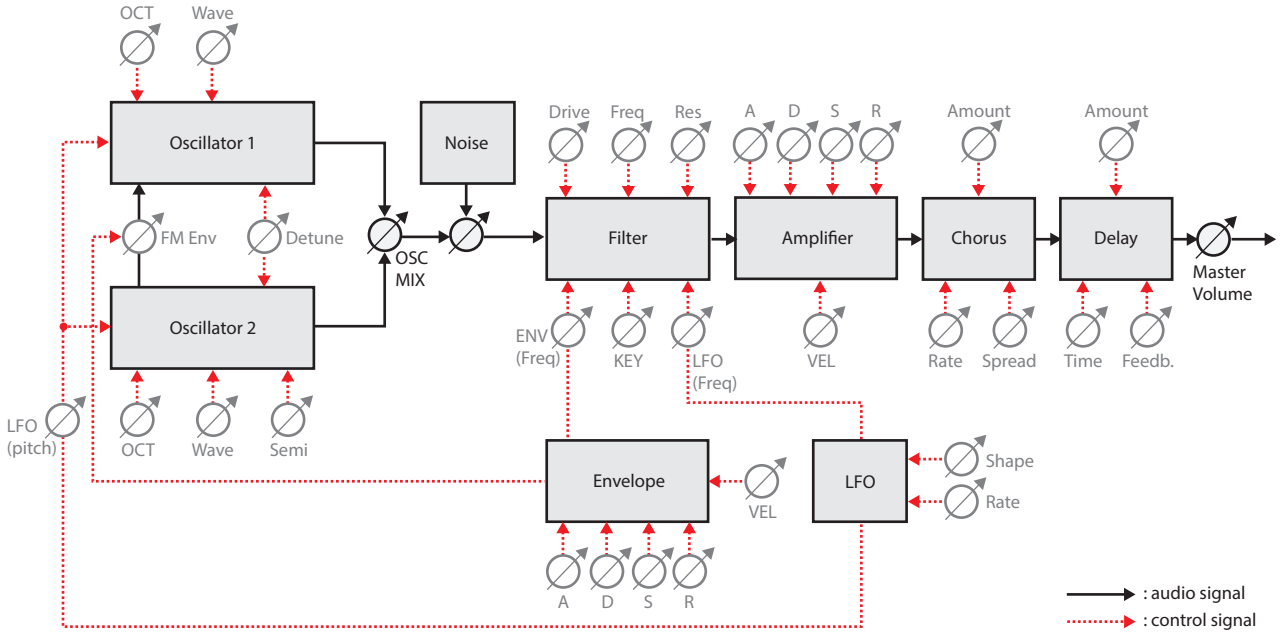


The Monotone front panel sections.

- **1. MIDI Note On LED.**
- **2. Patch Selector (for browsing, loading and saving patches).**
- **3. Master Volume.**
- **4. Voicing section and global controls.**
- **5. Oscillator section.**
- **6. Filter section.**
- **7. Amplitude Envelope.**
- **8. Chorus section.**
- **9. Delay section.**
- **10. Modulation section.**

Signal flow

The picture below shows the basic signal flow in Monotone:



Monotone signal flowchart.

- **The "heart" of Monotone is the Oscillator section, which generates the basic audio signal.**
 There are two oscillators in Monotone, that can produce a number of traditional "analog" waveforms - plus a Noise generator, which produces white noise. The two oscillators can be detuned, to create a fatter sound. Oscillator 2 can also frequency modulate Oscillator 1, for metallic/bell type sounds.
- **The signal from the Oscillator section is routed to the Filter.**
 The Filter in Monotone is a classic 24 dB/octave lowpass ladder filter, with overdrive control.
- **The signal from the Filter is routed to the Amplifier.**
 The Amplifier is controlled by a standard ADSR (Attack-Decay-Sustain-Release) envelope.
- **The signal from the Amplifier is then routed to the Chorus module.**
 This module generates a chorus effects, to make the sound fat and wide.
- **The signal from the Chorus is then routed to the Delay module.**
 Here you can add audio delay effects.
- **The remaining section in Monotone (Envelopes and LFO) can be used for modulating some Oscillator and Filter parameters.**

Playing and using Monotone

Loading and saving patches



Loading and saving patches is done in the same way as with any other internal Reason device, see "Loading patches" and "Saving patches" for details.

Global output controls



Master Volume

This is the main stereo output volume control.

Global performance and "play" controls



Portamento

Portamento makes the note pitch glide from the previous note to the new one, at the time set with the Portamento knob.

- **When On, the pitch will always glide between consecutive notes.**
- **In Auto mode, the pitch will glide between consecutive notes only when you play legato.**
 If you release the previous key before hitting the new key, there will be no portamento effect.

Retrig

→ **Click the Retrig button if you want to play Monotone and always retrigger the envelopes as soon as you play a new note.**
When Off, the envelopes will retrigger only if you have released the previous note before playing the new note.

Range

→ **Set the desired Pitch Bend range for the "Pitch" wheel with the up/down buttons, or by click-holding on the display and dragging up/down.**
Range: +/-24 semitones (+/-2 octaves) in steps of +/-1 semitone.

Pitch

The Pitch bend wheel can be used for bending note pitches up and down. Monotone also responds to Pitch Bend MIDI data from a connected MIDI master keyboard. You set the desired Pitch bend Range with the "Range" control above the Pitch bend wheel.

Mod

The Mod wheel can be used for controlling the Filter Frequency and LFO intensity in Monotone.

→ **Raise the FILT knob above the Mod wheel to set the Filter Frequency modulation amount.**

→ **Raise the LFO knob above the Mod wheel to set the LFO intensity modulation amount.**
Note that for the LFO modulation to work you need to already have some LFO modulation set in the Oscillator (see "LFO") and/or Filter (see "LFO") sections.

Panel reference

The Oscillator section



Here is where you choose oscillator waveforms and set the pitches for the two oscillators. You can also add noise and frequency modulate Oscillator 1 from Oscillator 2.

Waveform selector

→ **Turn the Wave knob to select one of four wave shapes.**

The wave shapes are:

- **Ramp**
 Also known as sawtooth. Generates a rich tone with both even and odd harmonics (overtones).
- **Square (Pulse in Oscillator 2)**
 The square wave has a symmetric square shape and contains only even harmonics. The Pulse wave is basically a square wave with non-symmetrical shape, i.e. a duty cycle that is not 50%. The Pulse wave generally sounds a little thinner than a perfect square wave.

- **Triangle**
 The Triangle wave only contains odd harmonics, and at lower intensities than the square wave overtones. This makes it sound a little "rounder" and with less bite than the square wave.
- **Sine**
 The Sine wave doesn't contain any overtones - only the fundamental. This makes it sound dull and makes it perfect as a sub bass an octave or two below another waveform in the other oscillator.

Oct

→ **Set the pitch in octave steps.**
Range: 5 octaves.

Osc Mix

→ **Set the mix of the Oscillator 1 and 2 signals.**

Noise

→ **Turn up the Noise knob to introduce white noise to the oscillator signal mix.**

Detune

→ **Change the pitch in steps of 1 cent (in opposite directions for the two oscillators).**
Range: +/- 50 cents (down or up half a semitone).

LFO

→ **Turn the LFO knob to introduce pitch modulation to both oscillators from the current setting of the LFO section (see "The LFO section").**

FM Env

→ **Turn the knob to have the oscillator 2 signal frequency modulate the oscillator 1 signal according to the current Envelope settings (see "The Envelope section").**
Range: 0% (no tracking (constant pitch)) to 100% (1 semitone per note).

Osc 2 Semi

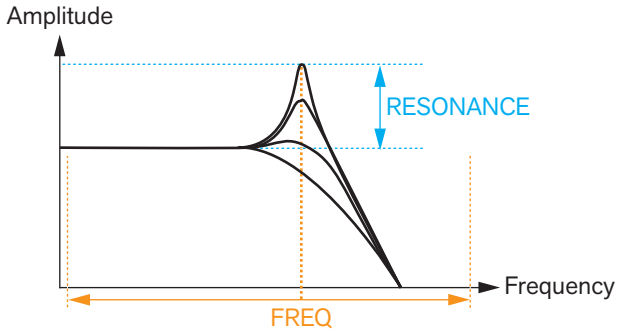
→ **Set the pitch of Oscillator 2 in semitone steps.**
Range: +/-12 semitones (two octaves).

The Filter section



The Filter in Monotone is a classic 24 dB/octave lowpass ladder filter. If you raise the Resonance high enough, the filter will start to self-oscillate.

The picture below shows the lowpass filter's basic characteristics at four different resonance levels:



Drive

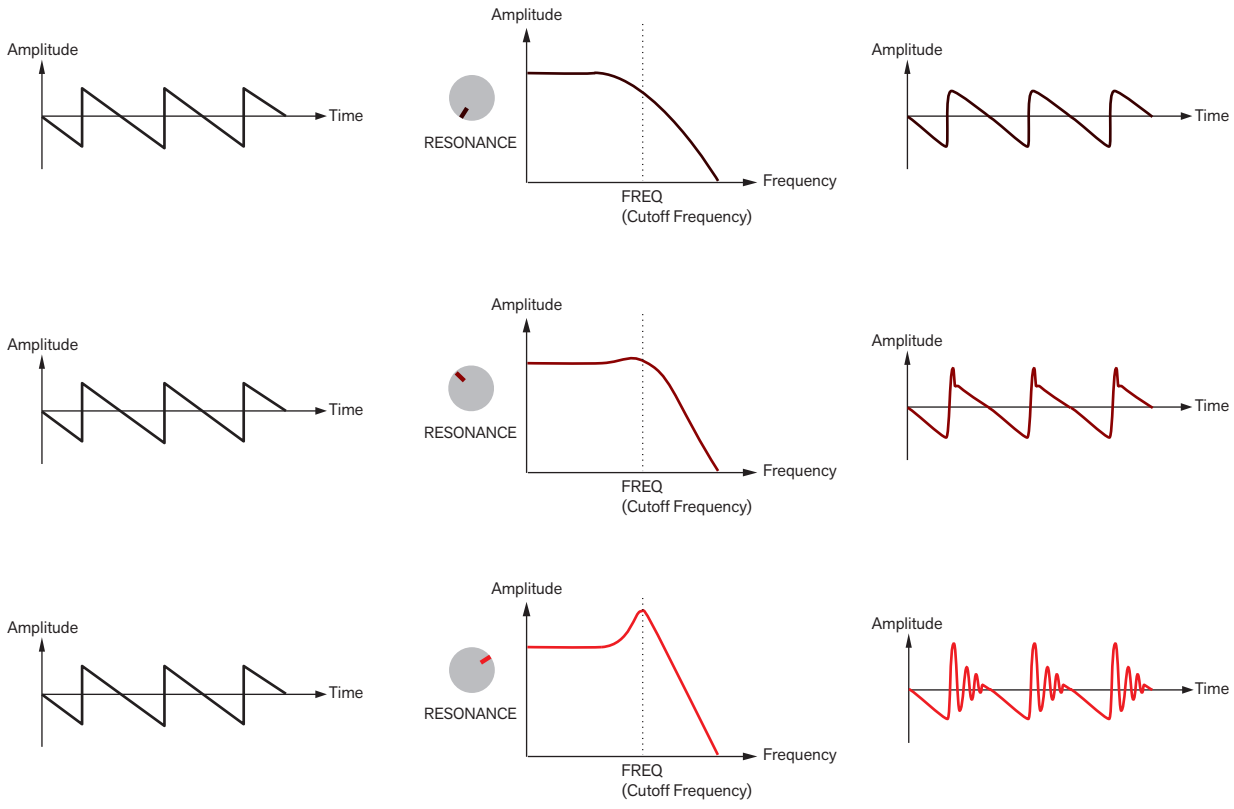
→ **Turn the Drive knob to amplify and introduce an overdrive type of distortion to the signal fed into the filter.**

Freq

→ **Set the cutoff frequency for the filter.**
The cutoff frequency is where the filter starts to cut out/dampen the higher frequencies of the signal.

Resonance

→ **Set the resonance amount.**
This controls the resonance peak level at the currently set cutoff frequency (see "Freq" above).
The picture below shows a ramp oscillator signal lowpass-filtered at three different Resonance levels:



! **Be careful when using high Resonance values as this could generate quite loud audio levels!**

Env

→ **Turn the Env knob to set the frequency modulation amount from the current settings of the Envelope (see "The Envelope section").**

Key

→ **Turn the Kbd knob to set the keyboard tracking amount.**
At 0% the filter cutoff frequency is static and doesn't track the keyboard at all.
At 100% the filter cutoff frequency tracks the keyboard 1 semitone per note.

LFO

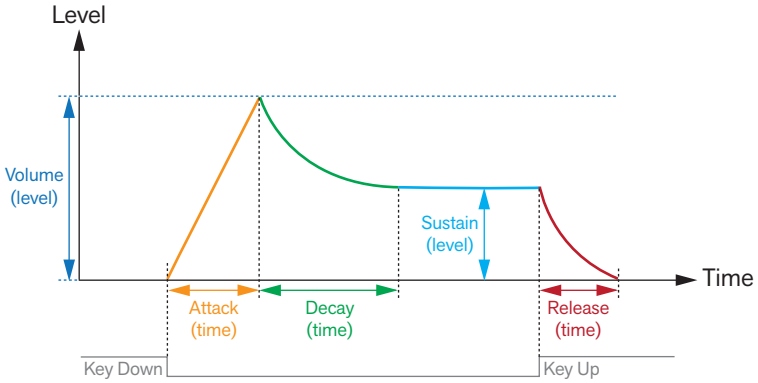
→ **Turn the LFO knob to set the frequency modulation amount from the current settings of the LFO (see "The LFO section").**

The Amplifier section



The Amplifier section contains a standard ADSR envelope, which controls the amplitude of the audio signal.

The picture below shows the various stages of the ADSR envelope:



The ADSR envelope stages.

A(ttack)

When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to max level. How long this should take, depends on the Attack setting. If the Attack is set to "0", maximum level is reached instantly. If the Attack value is raised, it will take longer time before the maximum level is reached.

D(ecay)

After maximum level has been reached, the level starts to drop. How long this should take is governed by the Decay parameter.

If you want to emulate the volume envelope of a note played on a piano for example, the Attack should be set to “0”, the Decay parameter should be set to a medium value and the Sustain level should be set to “0”, so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you raise the Sustain parameter.

S(ustain)

The Sustain parameter determines the level the envelope should rest at, after the Decay stage. If you set Sustain to full level, the Decay setting is of no importance since the volume of the sound is never lowered.

If you want to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack “0”) and stays there (Decay “0”), until the key is released and the sound instantly stops (Release “0”).

But often a combination of Decay and Sustain is used to generate envelopes that rise up to max level, then gradually decreases to finally land to rest on a level somewhere in-between zero and maximum level. Note that Sustain represents a level, whereas the other envelope parameters represent times.

R(elease)

The Release parameter works just like the Decay parameter, except it determines the time it takes for the volume to drop back to zero after you release the key.

Vel

- **Turn up the Vel knob if you want the maximum level to be controlled from Keyboard Velocity.**
 The harder you play, the louder the maximum volume.

Chorus



This is a stereo Chorus effect, which can be used for generating a fatter and wider sound.

Amount

- **Set the Dry/Wet amount of the chorus effect.**
 Set to 0% for a completely dry (unprocessed) signal.

Rate

- **Set the rate/speed of the chorus modulation.**

Spread

- **Set the stereo width of the chorus effect.**
 Set to 0% for a if you want the signal to be in mono.

Delay



This is a stereo delay, which generates delayed copies of the original signal.

Amount

- **Use this parameter to adjust the send level to the Delay effect.**
 Set to 0% for a completely dry (unprocessed) signal.

Time

The delay time is synced to the main sequencer tempo.

- **Set the sync division to the main sequencer tempo with the Time knob.**
 Range: 1/16, 1/8T, 1/8, 2/8T, 3/16, 1/4, 5/16, 4/8T, 7/16 and 2/4.

Feedback

- **Set the number of delay repeats.**

Ping Pong

- **Activate Ping Pong to have the delay repeats alternate between left and right in the stereo panorama.**

The LFO section



An LFO (Low Frequency Oscillator) is used for generating cyclic modulation. A typical example is to have an LFO modulate the pitch of a signal to produce vibrato, but there are also other applications for LFOs. The LFO section features an LFO which can be set to control Oscillator pitch (see "LFO") and/or Filter frequency (see "LFO").

Rate

- **Set the LFO rate/speed.**
 Range: 0.06-94.0 Hz

Shape

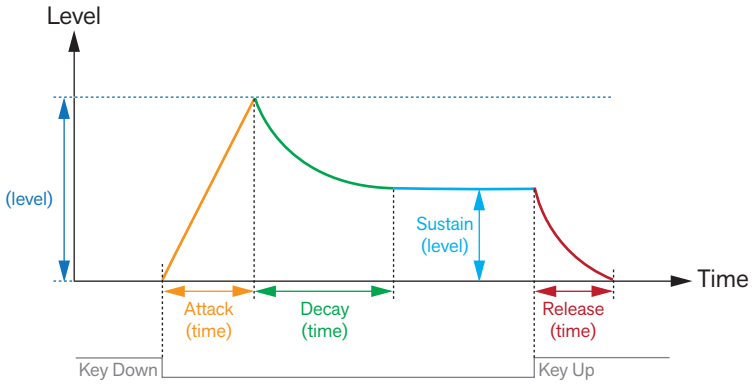
- **Turn the Shape knob to select one of three LFO wave shapes.**
 The wave shapes are: Sine, Triangle and Square.

The Envelope section



The Envelope section features a standard ADSR envelope, which can be used for controlling Oscillator Frequency Modulation (see "FM Env") and/or Filter Frequency (see "Env").

The various envelope stages work exactly like those of the Amplifier, see "The Amplifier section".



The ADSR envelope stages.

A(ttack)

- **Set the time it should take to reach from zero to maximum level.**

D(ecay)

- **Set the time it should take to go from maximum level to the Sustain level (see below).**

S(ustain)

- **Set the level the envelope should rest at, after the Decay stage (see above).**

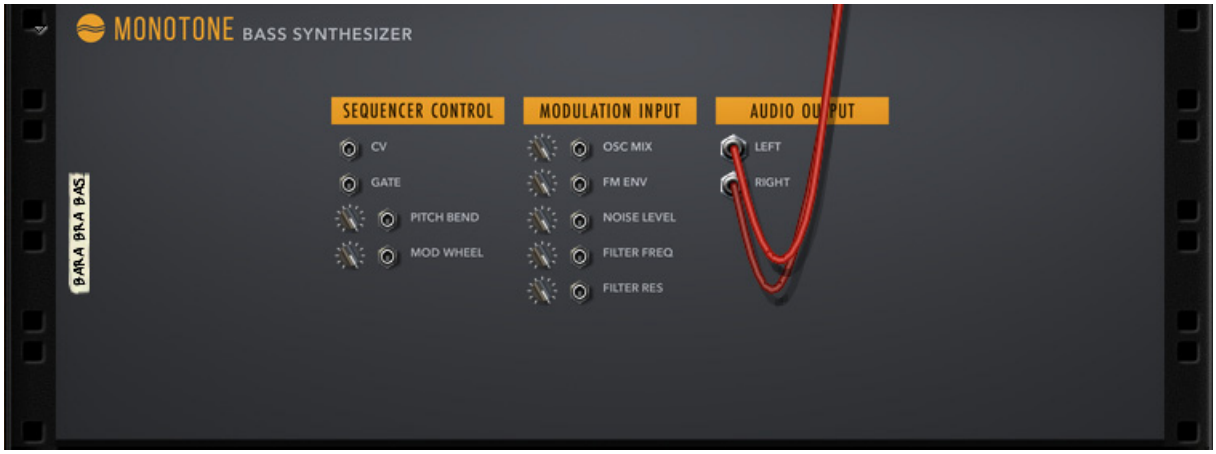
R(elease)

- **Set the time it should take to go from the Sustain level back to zero, after you have released the note.**

Vel

- **Turn up the Vel knob if you want the maximum level to be controlled from Keyboard Velocity.**
 The harder you play, the higher the maximum level.

Connections



! **Remember that CV connections are NOT stored in the Monotone patches! If you want to store CV connections between devices, put them in a Combinator device and save the Combi patch.**

Sequencer Control inputs

The Sequencer Control CV and Gate inputs allow you to play Monotone from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity. There are also inputs for modulating the Pitch Bend and Mod Wheel parameters.

Modulation inputs

These control voltage (CV) inputs and can be used for modulating the corresponding parameters from external modulations sources.

Audio Output

These are the main audio outputs. When you create a new Monotone device, these outputs are auto-routed to the first available outputs in the I/O device.

Chapter 20
ID8 Instrument Device

Introduction



The ID8 Instrument device is a synth module packed with great sounds - ideal for quickly creating nice complete arrangements. The sounds have been extracted from various Reason devices and ReFills to guarantee supreme audio quality.

The Sounds

The ID8 contains 36 presets divided into nine categories, with four sounds in each category. The categories are these:

- **Piano**
 The Piano category features a grand piano, an upright piano, a dance oriented piano sound and vibes.
- **Electric Piano**
 The Electric Piano category holds two classic electric piano sounds plus a digital FM type piano and a Clav.
- **Organ**
 The Organ category contains two classic tone-wheel organ sounds, one transistor organ sound and a pump organ.
- **Guitar**
 The Guitar category sports an acoustic steel string guitar, a clean electric guitar, a half-acoustic jazz guitar and a dulcimer.
- **Bass**
 The Bass category features one fingered and one picked electric bass, an acoustic upright bass and a synth bass.
- **Strings**
 The Strings category holds orchestral strings, arco strings, a small string section and a choir sound.
- **Brass-Wind**
 The Brass-Wind category features Fat Brass, Brass Section, French Horns and Flute.
- **Synth**
 The Synth category contains two classic monophonic synth lead sounds and two characteristic polyphonic pad sounds, one with fast attack and one with slow.
- **Drums**
 The Drums category sports four extensive combinations of drums and percussion instruments aimed at different musical styles. Each "drum kit" contains between 53 and 65 different instruments, so there is plenty to choose from!

See "Velocity mapping" for information about the velocity mapping of some of the sounds.

Using the ID8

Selecting Sounds

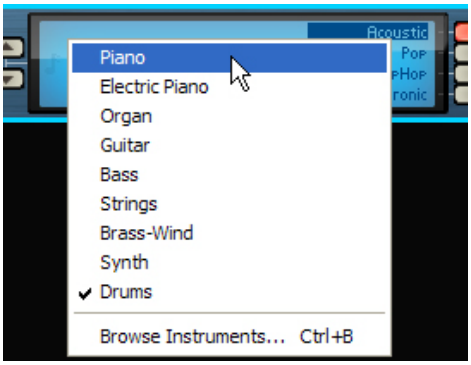
- **Select Category by clicking the Up/Down buttons to the left of the Display.**



- **Select Sound in the selected Category by clicking on any of the A-D buttons, or by clicking on the Sound name in the Display.**



- **Click on the Category name in the ID8 Display to bring up a pop-up where you can select Category or replace the ID8 device with another device.**



At the bottom of the pop-up, you can also choose “Browse Instruments...”. Selecting this allows you to replace the ID8 device with another instrument device and load a new sound in that device.

Controlling Sounds

Parameter knobs



Each of the Sounds in the ID8 have two preset parameters assigned to the Parameter 1 and 2 knobs. The parameter names are shown in the small displays to the right of the corresponding knobs, and are different depending on the selected Sound.

Pitch Bend and Mod Wheel



To the left on the ID8 front panel are the standard Pitch Bend and Mod Wheel. The Pitch Bend range is +/- 2 semitones and is the same for all sounds. The Mod Wheel is assigned a little differently depending on the selected Sound, but usually controls vibrato. In the Drums Category, however, the Mod Wheel has no effect, except on the Electronic Drums where it controls the cutoff frequency of a lowpass filter.

Performance Controllers

The sounds in the ID8 also respond to the standard Performance controllers "Sustain Pedal", "Aftertouch", "Expression" and "Breath Control". The parameter assignments can be a little different depending on selected sound. However, "Sustain Pedal" always controls sustain and "Expression" always controls volume.

Velocity mapping

A lot of the sounds in the ID8 are multi-sampled. They also have several velocity layers to faithfully reproduce the original instruments. Some of the sounds also use different types of samples for the highest velocity layer. This means that instead of just sounding louder, they will also sound different. For example, the Jazz Semi Guitar as well as the Finger, Pick and Upright basses have glissando or sliding notes in the highest velocity layer. The Arco Strings have pizzicato (picked) notes in the top velocity layer.

About saving edited Sounds

Since the ID8 is designed as a "preset" sound module, there is no dedicated function for saving edited Sounds. However, any edits you have made of the parameters in a Sound are automatically stored with the song (document) when you save the song.

- **You could also include one or several ID8 Instrument devices in a Combinator device and save the Combinator patch. Doing so will automatically store the settings of the ID8 Parameters in the Combinator patch. See "The Combinator" for more details.**

Chapter 21
Rytmik Drum Machine

Introduction

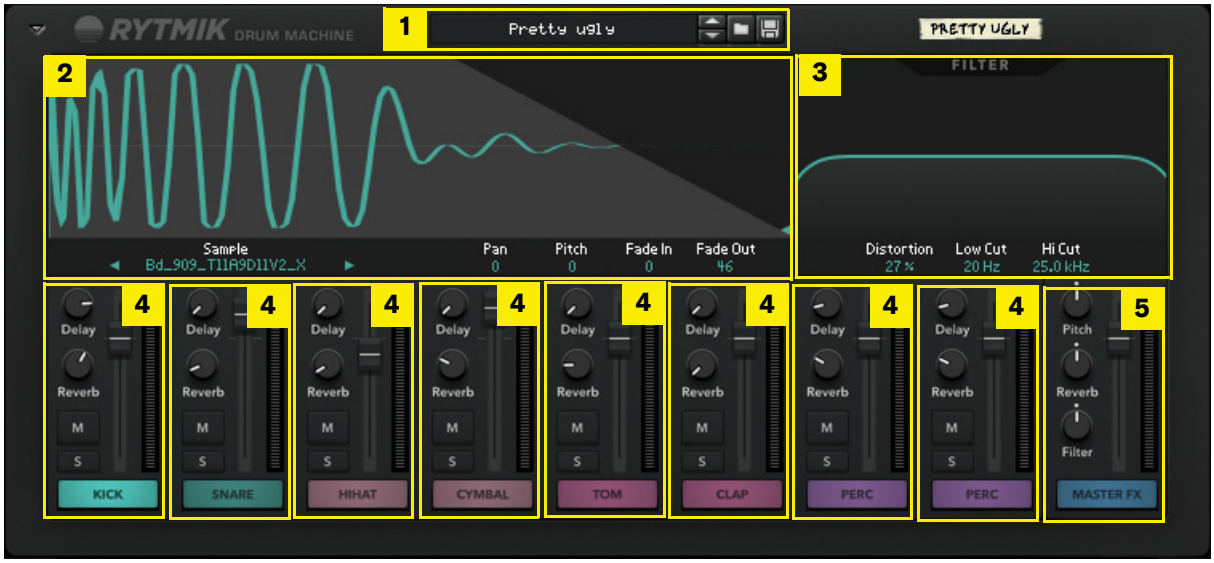


The Rytmik Drum Machine device is an eight-channel drum sample player. Rytmik features a Distortion effect and a Low Cut + Hi Cut filter per drum channel. There are also two send effects - Reverb and Delay - that are shared among the eight drum channels, plus a master section with a Master Compressor, Master Pitch and a Master Filter.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

Panel overview

The Rytmik front panel contains the following sections:

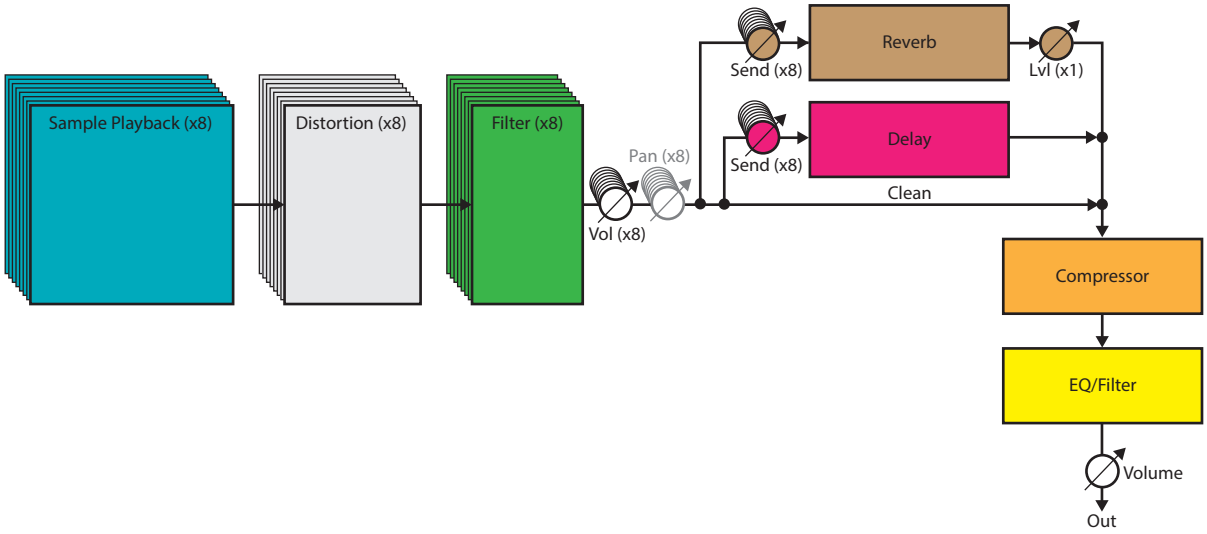


The Rytmik front panel sections.

- **1. Patch selector (for browsing, loading and saving patches).**
- **2. Sample playback and editing section (for the currently selected Drum Channel).**
- **3. Distortion and Low Cut + Hi Cut Filter (for the currently selected Drum Channel).**
- **4. Drum Channel sections (for playing back samples and for selecting Drum Channel to edit).**
- **5. Send Effects and Compressor section (global for all eight Drum Channels).**

Signal flow

The picture below shows the basic signal flow in Rytmik:



Rytmik signal flowchart.

- **Rytmik features eight Sample Playback engines (one per Drum Channel).**
 Each Sample Playback engine features a Sample section, where you can select any of the built-in samples. Here, you can also set the Sample Start, Sample End, Pan, Pitch, Fade In and Fade Out parameters.
- **The audio from the Sample Playback engines are then routed to a Distortion insert effect (one per Drum Channel).**
- **The signal is then routed to a Filter section (one per Drum Channel).**
 The signal can then be processed with a Low Cut and Hi Cut filter.
- **The signal from each Drum Channel can then be panned in stereo - and have its own individual volume.**
- **Each Drum Channel can also use the two global Send Effects (Reverb and Delay).**
 The signal levels to each of these send effects can be set individually for each Drum Channel.
- **Finally, the signals of all eight Drum Channels, plus the Send Effects are mixed and output as a stereo signal via a Master Compressor and a Master Filter.**
 It's also possible to output the desired Drum Channel signals individually via separate audio outputs - if you want to process the signals outside of the Rytmik device. If you do that, the signal will be output after the Filter section of the Drum Channel, bypassing the Send Effects and the Master FX.

Global controls

Loading and saving patches



Loading and saving patches is done in the same way as with any other internal Reason device, see "Loading patches" and "Saving patches" for details.

The Drum Channel sections



Auditioning samples

→ **Click a Drum Channel button to play back the sample of the corresponding Drum Channel.**
By clicking the Drum Channel button you also automatically select the Drum Channel (see below).



- **If you are using a MIDI Keyboard/On-screen Piano Keys you can play back the samples of the Drum Channels from Key C0 to G0.**

Selecting a Drum Channel

→ **Click a Drum Channel button to select the desired Drum Channel:**



Muting and soloing Drum Channels

→ **Click the M(ute) or S(olo) button in a Drum Channel to mute or solo the desired Drum Channel:**



Setting the Drum Channel volumes

→ **Drag the Volume slider up/down - or just click - to set the volume of a Drum Channel:**

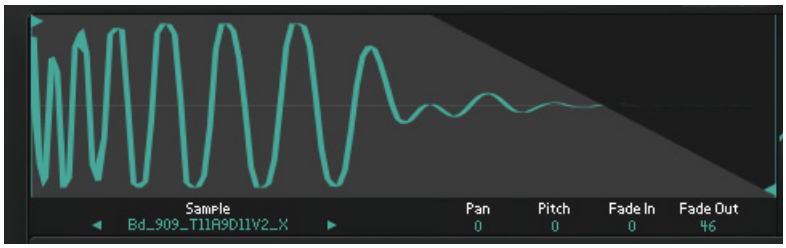


Setting the Send Effect levels

→ **Turn the Delay and Reverb knobs to set the send levels from the Drum Channel to the respective Send Effect:**
See "The Master FX section" for more information about the Send Effects.



The Sample Playback section



The Sample Playback section contains all sample related controls and parameters for the currently selected Drum Channel. The currently selected Drum Channel is indicated by the lit Drum Channel selection button (see "Selecting a Drum Channel").

The Sample Playback section features the following parameters and controls:

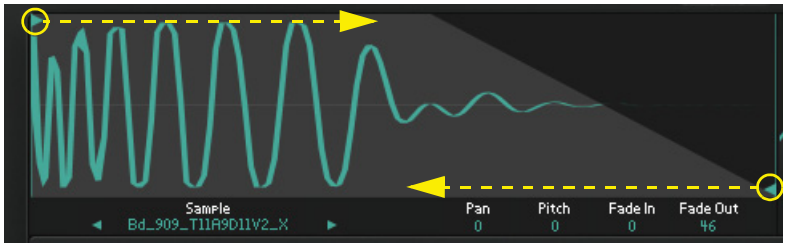
Selecting Samples



→ **Select and load a sample either by clicking the triangular arrow buttons on either side of the sample name - or by clicking the sample name and selecting from the pop-up menu.**
The pop-up menu features eight sub-groups with different types of samples.

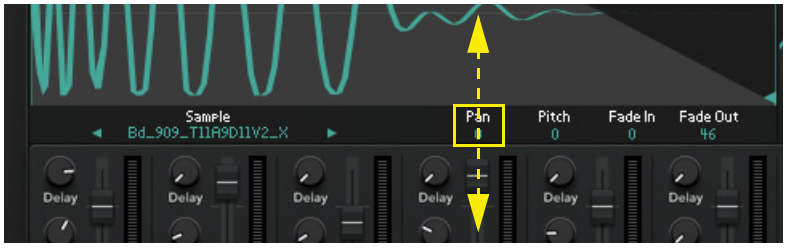
! **All samples in Rytmik are embedded in the device itself, so when you save a Rytmik patch the samples are always included (as opposed to other sampler devices in Reason).**

Setting the Sample Start and End



- **Click and drag the Sample Start handle sideways to change the where in the sample playback should begin.**
- **Click and drag the Sample End handle sideways to change the where in the sample playback should stop.**

Setting the Panning



- **Click and drag up/down in the Pan box to place the sample in the stereo panorama.**
 Range: 100L to 100R.

Setting the Pitch



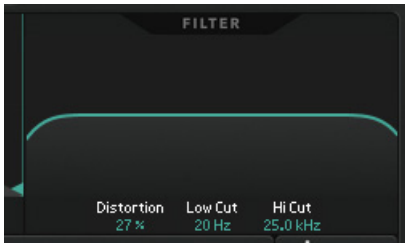
- **Click and drag up/down in the Pitch box to set the pitch of the sample.**
 Range: +/- 1200 cents.

Setting Fade In and Fade Out



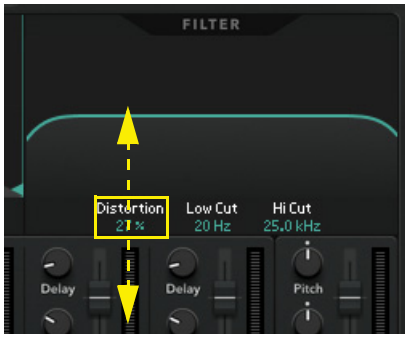
→ **Click and drag up/down in the Fade In and/or Fade Out boxes to apply a fade in/out of the sample.**
Any fade in/out is shown graphically in the sample display.

The Insert Effects section



The Insert Effects section consists of a Distortion effect and a Low Cut and Hi Cut Filter.

Distortion

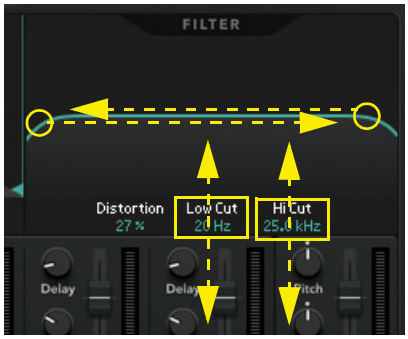


The Distortion effect is a transistor type of distortion.

→ **Click and drag up/down in the Distortion box to adjust the input gain to the distortion effect.**
A high value will overdrive the pre-amp and generate more distortion.

Range: 0-100%, where “0” is completely dry/off.

Low Cut and Hi Cut Filter



The Filter features a Low Cut (Highpass) and a Hi Cut (Lowpass) filter. Perfect for removing any rumble (Lo Cut) and/or hiss (Hi Cut), for example.

- **Click and drag up/down in the Low Cut box to set the cutoff frequency for the highpass filter.** Alternatively click the left part of the filter curve in the display and drag sideways.

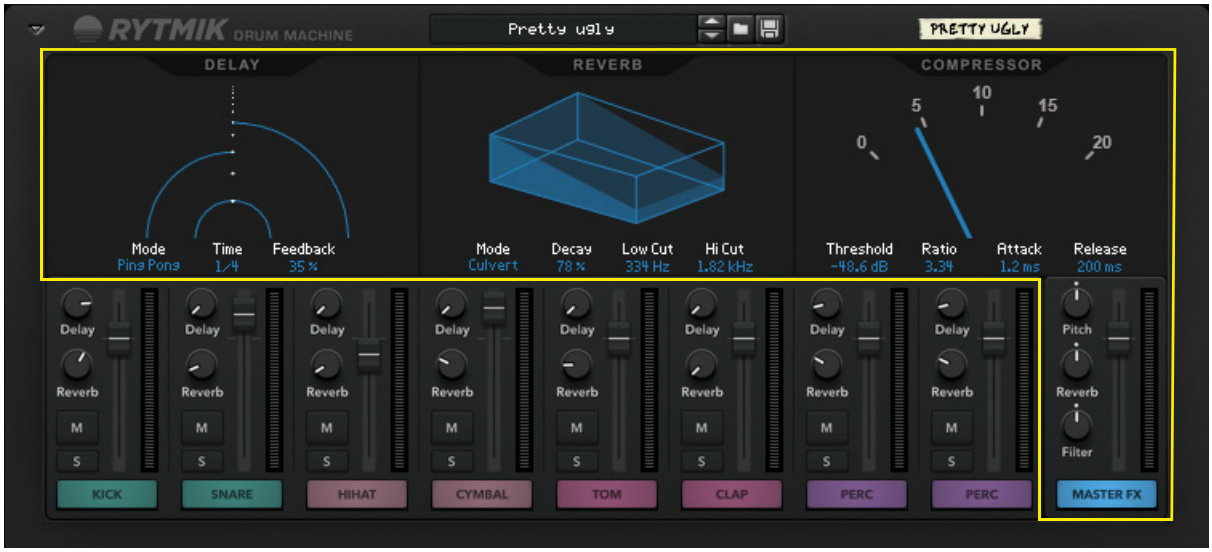
 Range: 20 Hz to 25 kHz.

- **Click and drag up/down in the Hi Cut box to set the cutoff frequency for the lowpass filter.** Alternatively click the right part of the filter curve in the display and drag sideways.

 Range: 20 Hz to 25 kHz.

! **Note that the Low Cut and Hi Cut cutoff frequencies can also be on opposite sides of each other, which means that the level of the sample could eventually drop to zero with extreme settings.**

The Master FX section



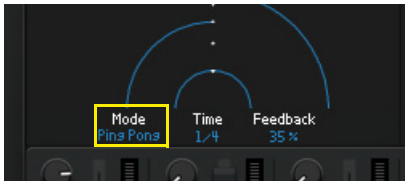
The Master FX section features controls for the Delay and Reverb send effects, as well as for the master bus Compressor. The Send Effects can be used by all Drum Channels simultaneously, and the effects are active simultaneously.

Delay



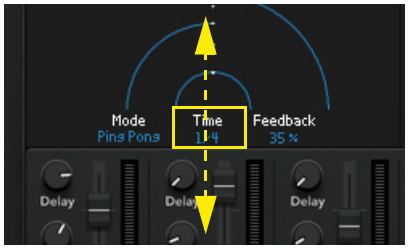
This is a delay with two different modes - Stereo and Ping Pong. The delay repeats are tempo synced to the main sequencer and you can select the desired time division.

Mode



→ **Click the Mode box and select the desired mode from the pop-up.**
"Stereo" repeats the delay in stereo in a fixed centered position.
"Ping Pong" repeats the delays, alternating between the left and right channels.

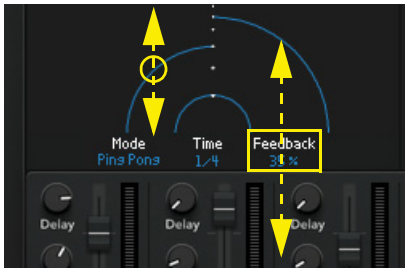
Time



→ **Click and drag the Time box up/down to set the time division of the tempo-synced delay repeats.**
The tempo is hard-wired to the main sequencer tempo.

Time divisions: 1/1, 1/2D, 1/1T, 1/2, 1/4D, 1/2T, 1/4, 1/8D, 1/4T, 1/8, 1/16D, 1/8T and 1/16

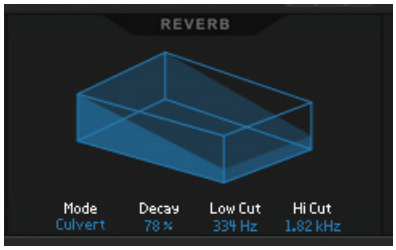
Feedback



→ **Click and drag the Feedback box up/down to set the number of delay repeats.**
Alternatively, click and drag in the display to set the Feedback amount.

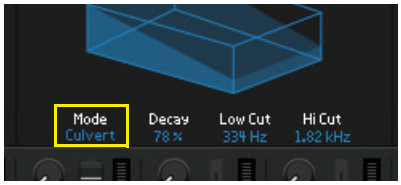
Range: 0-100%, where "0" is one single delay repeat.

Reverb



This is a stereo reverb with six different Modes (reverb algorithms).

Mode

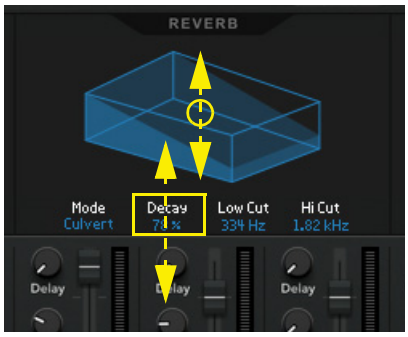


→ **Click the Mode box and select the desired reverb algorithm from the pop-up.**

The following reverb types can be selected:

- **Room**
- **Large Room**
- **Culvert**
- **Plate**
- **Gated**
- **Hall**

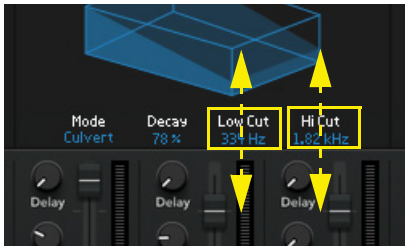
Decay



→ **Click and drag the Decay box up/down to set the length of the reverb effect.**
Alternatively, click and drag in the display to set the decay length.

The Decay amount is also shown graphically in the display.

Low Cut and Hi Cut

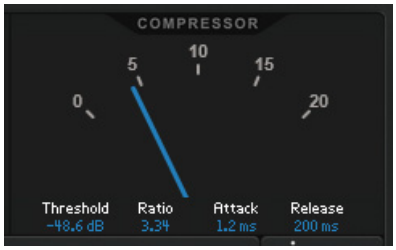


This is essentially a combination of a highpass and a lowpass filter.

- **Raise the Low Cut value to cut the low frequencies of the reverb signal and make the reverb effect less "muddy".**
 Range: 20 Hz to 25 kHz.
- **Lower the Hi Cut value to cut off the high frequencies of the reverb, thereby creating a smoother, warmer effect.**
 Range: 20 Hz to 25 kHz.

! **Note that the Low Cut and Hi Cut cutoff frequencies can also be on opposite sides of each other, which means that the reverb level could eventually drop to zero with extreme settings.**

Compressor



This is a stereo compressor, which acts on the total mix of all Drum Channels. It is always active, but if you don't want any compression effect you can set the controls so that no compression is produced. The gain reduction is shown by the meter.

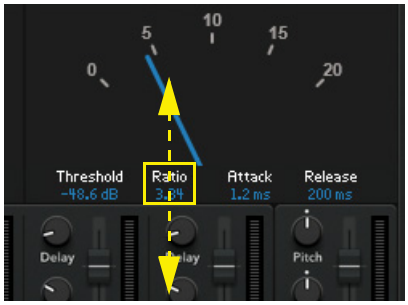
Threshold



This is the threshold level above which the compression sets in. Signals with levels above the threshold will be affected, signals below it will not. In practice, this means that the lower the Threshold setting, the more the compression effect.

- **Click and drag the Threshold box up/down to set the threshold level.**
 Range: -60 dB to 0 dB
 - **For no compression effect at all, set the Threshold to "0 dB".**

Ratio

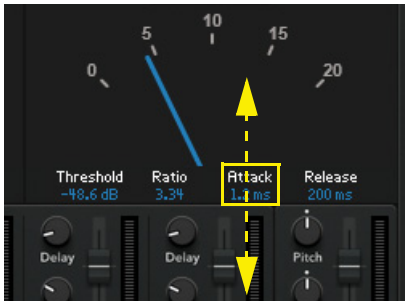


This specifies the amount of gain reduction applied to the signal above the set threshold. A high Ratio value makes the sound less dynamic and more "even" in level.

- **Click and drag the Ratio box up/down to set the compression ratio.**
 Range: 1:1 to 20:1
- **For no compression effect at all, set the Ratio to "1.00".**

Attack

This governs how quickly the compressor applies the gain reduction when signals rise above the set Threshold (see above). If you raise the Attack value, the response will be slower, allowing more of the signal to pass through the compressor unaffected. Typically, this is used for preserving the attacks of the sounds.



- **Click and drag the Attack box up/down to set the compressor attack time.**
 Range: 0-200 ms

Release

When the signal level drops below the set Threshold (see above), this determines how long it takes before the compressor lets the sound through unaffected. Set this to short values for intense, "pumping" compressor effects, or to longer values for a smoother change of the dynamics.



- **Click and drag the Release box up/down to set the compressor release time.**
 Range: 0-200 ms

Master Pitch



→ **Turn the Master Pitch knob to adjust the pitches of all Drum Channels equally.**
Range: +/-1200 cents.

Master Reverb



→ **Turn the Master Reverb knob to adjust the Reverb return level.**
Range: +/-100%.

! **Note that this control is bipolar, i.e. you could attenuate or amplify the reverb return level. This means that if any of the Reverb Amount knobs for the Drum Channels are < 0 dB, raising the Master Reverb knob to a positive value will increase the Reverb level for these Drum Channels. The level can never exceed 0 dB, though.**

Master Filter



The Master Filter is a combined highpass and lowpass filter, which can be used for cutting out low or high frequencies in the total mix signal. At the default 0% setting the output signal is completely unaffected (not filtered at all).

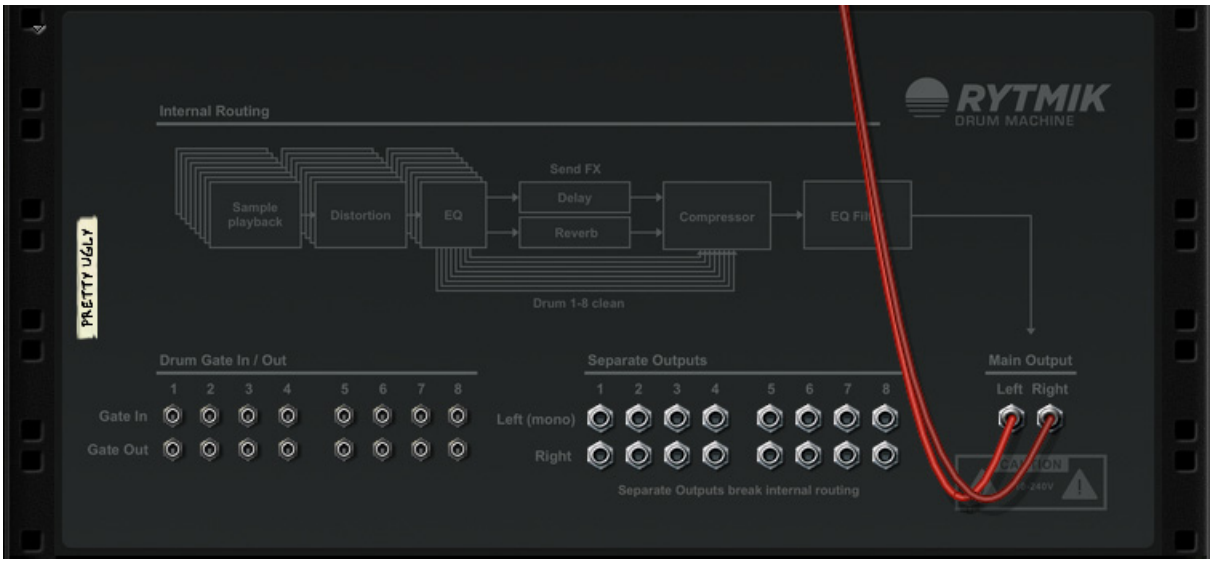
→ **Turn the Master Filter knob to adjust the Low Cut and Hi Cut effect.**
Range: +/-100%.

Master Volume



→ **Drag the Master Volume slider up/down - or just click - to set the output volume of the total mix.**
Range: -inf to 6.00 dB

Connections



! **Remember that CV connections are NOT stored in the Rytmik patches! If you want to store CV connections between devices, put them in a Combinator device and save the Combi patch.**

Drum Gate In/Out

The Drum Gate inputs allow you to play Rytmik from another CV/Gate device. The Drum Gate inputs respond to "Note On/Off" along with Velocity.

The Drum Gate outputs allow you to control other CV/Gate equipped devices from Rytmik. The Drum Gate outputs send out "Note On/Off" along with Velocity.

Separate Outputs

The separate outputs can be used for routing individual Drum Channels to external destinations, for further processing.

! **Note that Drum Channels routed via separate outputs are automatically disconnected from the Master FX section. Note, though, that the signal could still be sent to (and heard via) the Send Effects (Delay and Reverb) on the Main Outputs.**

Main Output L & R

These are the main audio outputs. When you create a new Rytmik device, these outputs are auto-routed to the first available outputs in the I/O device.

Chapter 22
Radical Piano

Introduction



Radical Piano is designed to be a straightforward, awesome sounding and very flexible piano. Radical Piano combines sample playback technology with physical modeling to give you great sound quality and seamless dynamic response as well as great freedom to tweak your sounds.

The combination of sample playback and physical modeling also makes it possible to keep each piano sound set down to a minimum size. This allows for very quick loading times when you switch between instruments.

Radical Piano also features sympathetic resonance, which means that any undamped strings will ring along with the played notes (strings), just like on acoustic pianos. This makes Radical Piano sound extremely realistic and alive. There are also a number of other controls for further shaping the sound the way you want it.

As a bonus, we also included an audio input so you can route external audio and process it in Radical Piano!

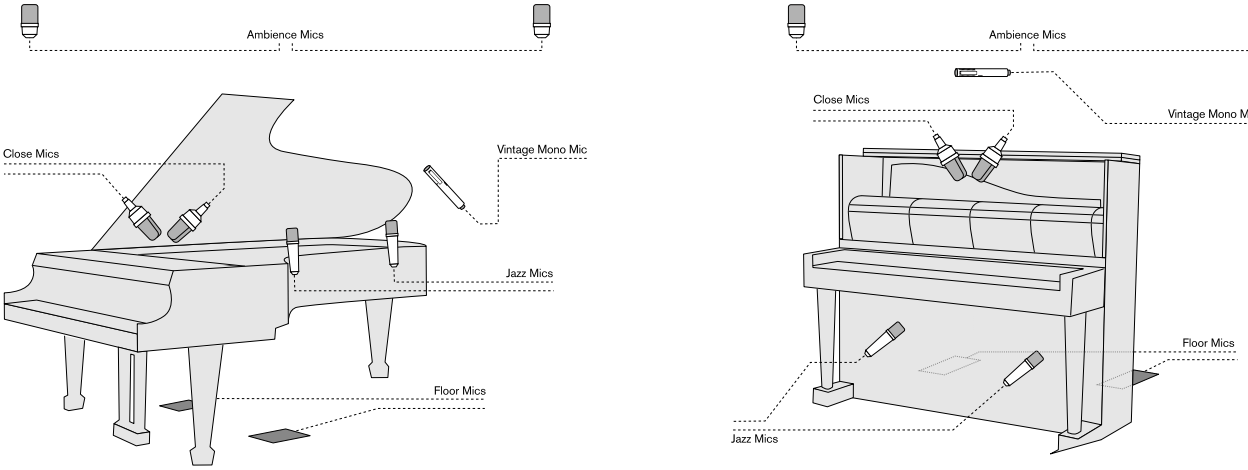
! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

The pianos

Radical Piano holds complete sound sets recorded from these three pianos:

- **Home Grand**
 A Bechstein grand piano with a nice “not perfectly tuned” home grand character.
- **Deluxe Grand**
 A Steinway Model D grand piano - one of the greatest grand pianos out there. This particular one belongs to Sveriges Radio (Swedish Radio Ltd).
- **Upright**
 A Futura upright piano with a distinct “living room” character.

The microphone configurations



The microphone configurations for the grand pianos and the upright piano respectively.

The pianos were recorded using up to nine microphones per instrument, placed at various critical positions inside and outside of the pianos. The different microphone recordings were then stored in Radical Piano as separate sound sets.

The following microphone configurations were used:

- **Vintage Mono**
 A single microphone placed outside the waist of the grand piano (or behind the upright piano). For the Steinway grand piano we used an old school mono ribbon mic with vintage characteristics and a narrow frequency response with the emphasis in the mid range. For the Futura upright piano we used a vintage tube microphone.
- **Ambience**
 Two microphones in AB configuration* placed quite far away from the piano to capture the room ambience.
- **Floor**
 Two pressure zone microphones that lay flat on the floor just behind the front legs of the grand piano (and behind the upright piano). They add depth and richness to the sound and are best used as a complement to the other mics.
- **Jazz**
 Two microphones in AB configuration* placed just outside/in front of the piano. This gives a full bodied sound with a wide stereo image and a less pronounced attack.
- **Close**
 Two microphones in XY configuration** placed close to the hammers. The close mics produce a distinct sound with a sharp attack, ideal for uptempo pop/rock.

* AB configuration: Two mics in stereo configuration placed several feet apart and tilted slightly away from each other.

** XY configuration: Two mics in stereo configuration placed close together in 'V' shape at a 90° coincidence.

Using Radical Piano

Loading and saving patches

Loading and saving patches is done in the same way as with any other internal Reason device, see "Loading patches" and "Saving patches" for details.

Selecting piano sound sets

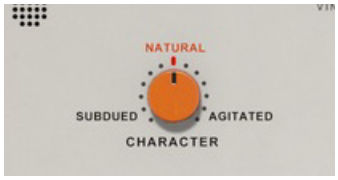
A patch in Radical Piano can consist of a mix between two piano sound sets. The mix could be between two sound sets from the same piano, or from different pianos. You could, for example, blend a Close mic'ed upright piano with the Floor microphones from a grand piano to create your own custom piano sound. The piano sound sets can be selected in the Piano Select section:



The Piano Select section.

1. **Select desired piano sound set(s) by clicking the corresponding LED button(s).**
 You can select one sound set to the left of the Blend knob and one to the right.
2. **Set the mix between the sound sets with the Blend Microphones knob.**
 If you only want to use a single sound set for your sound, set the Microphone Blend knob to min or max.

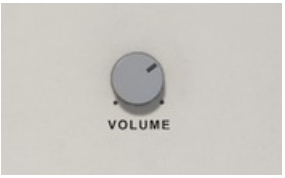
Character



→ **Set the character of the sound with the Character knob.**
Range: Subdued to Agitated, in 24 steps, with natural sound at the 12 o'clock position.
Subdued produces a warm and mellow tone whereas Agitated generates a brighter and significantly more pronounced tone.

! **Changing the Character value temporarily mutes the audio outputs.**

Volume



The master volume control for Radical Piano.

Velocity Response



Most sample-based piano instruments and sound libraries on the market use a predefined number of velocity layers. Depending on how soft or hard you play the keys, samples from a specific velocity layer play back. Due to memory limitations, the number of velocity layers aren't often that many. This can make the velocity response feel and sound unnatural. Thanks to the combination of samples and physical modeling in Radical Piano, all sound sets feature very wide and completely seamless velocity ranges.

With the Velocity Response knobs you can tailor the dynamic response of your piano sound.

- **With the High knob you set the timbre for the highest velocity.**
 Note that the High parameter can go far beyond the natural range of an acoustic piano, which is great for experimental sounds.
- **With the Low knob you set the timbre for the lowest velocity.**
 With the Low knob set to zero (marked with an 'S') playing really soft won't play back any sound at all. This can be useful if you, for example, want to hold down a chord and then play other keys to introduce the sympathetic resonance effect, see "Resonance".
- **With the Curve knob you set the shape of the velocity curve - from exponential, via linear to logarithmic.**
 Set this parameter where it feels the best to play. There is no "perfect" position since most MIDI keyboards respond differently to velocity.

- **If you want a natural dynamic range, set the Low knob to around the 9 o'clock position and the High knob to around the 12 o'clock position. Adjust the Curve setting to your liking.**
- **If you want a dynamic range that stretches beyond the range of an acoustic piano, set the Low knob to zero and the High knob past the 12 o'clock position.**
- **If you want a static response (with the same timbre no matter how soft or hard you play), set the Low knob to max and the High knob to zero. Note that there will still be some velocity sensitivity left for controlling the volume.**

Tune



Cent

→ **Set the overall master tune of your sound with the Cent knob.**
Range: +/-1 semitone (+/-100 cents).

Drift

The Drift parameter can be used for introducing a slow irregular pitch variation to your sound. It's perfect for adding kind of a "scary" or melancholic touch to your piano sound.

Sustain



The Sustain parameter is a special feature in Radical Piano. It lets you control the piano sustain continuously from pedal up to pedal down. As on acoustic pianos, the sustain pedal is not either "on" or "off - it can be "somewhere in between" as well. The Sustain function in Radical Piano simulates this behavior.

The Sustain parameter can be controlled either from the Pedal LED strip control on the front panel or from a Sustain pedal connected to the Sustain Pedal input of your MIDI master keyboard.

- **When you use a standard (switch type) sustain pedal connected to a standard sustain pedal input on your MIDI keyboard, this will control the Sustain function in Radical Piano as either Off ('0') or On ('127').**
 You could record using the standard sustain pedal and then manually edit the Sustain Pedal performance controller data in the note clip in Reason afterwards and adjust the "in between" Sustain levels.

! **The Sustain parameter value (and LED bar) will always adjust to the latest incoming Sustain Pedal data, be it from the Pedal LED strip control or from a sustain pedal connected to your MIDI keyboard.**

Resonance



Sympathetic resonance is a physical phenomenon that can occur in acoustic instruments, like in pianos for example. It means that any undamped strings will ring along with the played strings. For example, if you play a key with the sustain pedal down, all other strings in the piano will also vibrate at various intensities. Similarly, if you hold down a number of keys (so that the dampers are off the strings) and then play additional keys, the strings for the held keys will resonate.

With the Resonance controls you set the amount of sympathetic resonance in your piano sound.

Level

- **Set the amount of overall sympathetic resonance in your sound.**

Release Time

- **Set the time it should take for the sympathetic resonance to fade to silence.**

Envelope



Radical Piano features a special type of envelope generator which is used for shaping the character of the piano sound.

Attack

- **Set the attack time for the piano sound, from immediate to (unnaturally) slow.**
 The range is 0-200 ms.

Decay Curve

- **Set the shape of the decay curve.**
 This control determines how the sound should decay when you play and hold the keys.
 The range is from exponential, via linear, to logarithmic. Exponential settings will make the sound decay faster, which simulates a piano with little body sustain. Logarithmic settings makes the sound sustain more slowly and simulates a piano with a lot of body sustain.

Release

→ **Set the time it should take for the sound to fade to silence once you release the keys.**
This simulates the behavior of the dampers. For example, worn out dampers could result in somewhat longer release times.

Mechanics



The Mechanics section features controls for the mechanical noise.

Key Down

- **Key Down controls the level - and character - of the noise that occurs when the keys are pressed/hit.**
At the 12 o'clock position the noise is the most natural. Above the 12 o'clock position the noise is more pronounced and below the 12 o'clock position the noise is suppressed.

Key Up

- **Key Up controls the level of the noise that occurs when the keys are released and the hammers and dampers return to their initial positions.**
At the 12 o'clock position the noise level is natural. Above the 12 o'clock position the noise is louder and below the 12 o'clock position the noise is quieter.

Pedal

- **Pedal controls the level of the noise that occurs when you press and release the sustain pedal.**
At the 12 o'clock position the noise level is natural. Above the 12 o'clock position the noise is louder and below the 12 o'clock position the noise is quieter.

EQ



The built-in equalizer is a powerful 3-band EQ with gain controls for the Low, Mid and High bands. The EQ characteristics have been fine tuned and optimized for piano sounds. The gain range is +/-18dB for each of the bands, which makes it easy to quickly achieve great sonic results.

The EQ can be switched on/off by clicking the LED button at the top.

Ambience



The Ambience section features four different reverb types and a Level control. The reverb types are:

- **Small Room**
 This simulates the acoustic reflections in a small room.
- **Large Room**
 This simulates the acoustic reflections in a large room.
- **Hall**
 This simulates the acoustic reflections in a medium size hall.
- **Theater**
 This simulates the acoustic reflections in a large hall/theater.

Output



Comp(ression)

This controls the amount of compression of your piano sound.

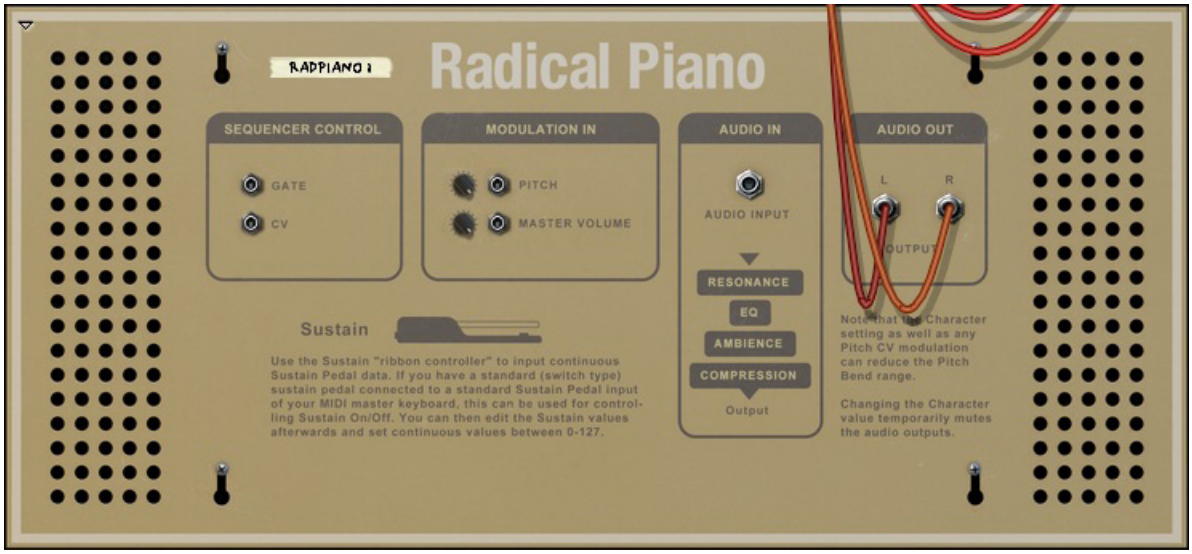
Width

This lets you set the stereo width of the piano sound.

! **Note that the Width control does not have any effect on the sound if you use only the "Vintage Mono" piano sound set(s), see "Selecting piano sound sets".**

Connections

! **Remember that CV connections will not be stored in the Radical Piano patch!**



Sequencer Control

The Sequencer Control CV and Gate inputs allow you to play Radical Piano from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity.

Modulation In

These control voltage (CV) inputs (with associated trim pots) can modulate following parameters in Radical Piano:

- **Pitch**
 The Pitch can be modulated at a maximum range of +/-1 octave.

! **Note that +/- 1 octave is the maximum range a piano sound can be pitch shifted in Radical Piano. This assumes that no Pitch Bend performance controller is used (see "Pitch Bend") and that the Character knob is set to Natural (see "Character").**

- **Master Volume**

Audio In

Route an external audio signal to this input to process it with the Resonance, EQ, Ambience and Compression effects in Radical Piano.

► **Routing a vocal signal and processing it with the sympathetic resonance effect (with the sustain pedal down) could generate really interesting results. It would be like singing into a piano body!**

Audio Out

These are the main audio outputs. When you create a new Radical Piano device, these outputs are auto-routed to the first available outputs in the I/O device.

Additional external control

The Radical Piano responds to the following standard Performance Controllers:

- **Pitch Bend**
 Radical Piano responds to Pitch Bend data from the pitch bend control of your MIDI master keyboard. The range is fixed at +/-7 semitones.

! **Note that the Character setting (see "Character") as well as any Pitch CV modulation (see "Pitch") can reduce the Pitch Bend range.**

- **Sustain Pedal**
 If you have a standard (switch type) sustain pedal connected to a standard Sustain Pedal input of your MIDI master keyboard, this can be used for controlling Sustain On/Off. You can then edit the Sustain values in your note clips in the sequencer afterwards and set continuous values all the way between 0-127, see "Sustain" for more details.

Chapter 23
Klang
Tuned Percussion



Introduction



The Klang Tuned Percussion instrument features an assortment of high-quality multi-sampled tuned percussion instruments - perfect for any music style. Each of the multi-sampled instruments can also be tailored and processed in the high-quality filter, amp, delay and reverb sections.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Panel overview

The Klang front panel contains the following sections:

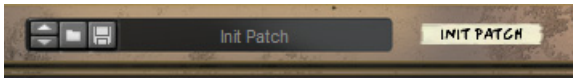


The Klang front panel sections.

- **1. Patch Selector (for browsing, loading and saving patches).**
- **2. Master Volume.**
- **3. Global performance and "play" controls.**
- **4. Instruments section.**
- **5. Filter section.**
- **6. Amp Envelope section.**
- **7. Delay effect section.**
- **8. Reverb effect section.**

Using Klang

Loading and saving patches



Loading and saving patches is done in the same way as with any other internal Reason device, see "Loading patches" and "Saving patches" for details.

Global performance and "play" controls



Note

The Note LED lights up each time Klang receives a MIDI Note On.

Range

→ **Set the desired Pitch Bend range for the "Pitch" wheel with the up/down buttons, or by click-holding on the display and dragging up/down.**
Range: +/-12 semitones (+/-1 octave) in steps of +/-1 semitone.

Pitch

The Pitch bend wheel can be used for bending note pitches up and down. Klang also responds to Pitch Bend MIDI data from a connected MIDI master keyboard. You set the desired Pitch bend Range with the "Range" control to the right of the Mod wheel.

Mod

The Mod wheel can be used for controlling the three predefined parameters to the right of the Mod wheel:

- **S. Start**
 Here you set how/if the Mod wheel should affect the Sample Start position of the currently selected instrument. The parameter is bi-polar, with zero modulation at the 12 o'clock position. A negative value moves the sample start back and a positive value moves it forward.

! **Note that if the S.Start parameter in the Instruments section (see "S. Start (Sample Start)") is at 0 ms, the sample start cannot be moved back any further. Similarly, if the S.Start parameter in the Instruments section is at 150 ms, the sample start cannot be moved forward any further.**

- **F. Freq**
 Here you set how/if the Mod wheel should affect the Filter Cutoff parameter. The parameter is bi-polar, with zero modulation at the 12 o'clock position.

! **Note that if the Cutoff parameter in the Filter section (see "Cutoff") is at 20 Hz, the frequency cannot be lowered any further. Similarly, if the Cutoff parameter in the Filter section is at 25 kHz, the frequency cannot be raised any further.**

- **Level**
 Here you set how/if the Mod wheel should affect the Master Volume parameter. The parameter is bi-polar, with zero modulation at the 12 o'clock position.

! **The volume can never be modulated louder than the maximum +12 dB limit of the Master Volume control.**

Panel controls

The Instruments section



Instrument selector



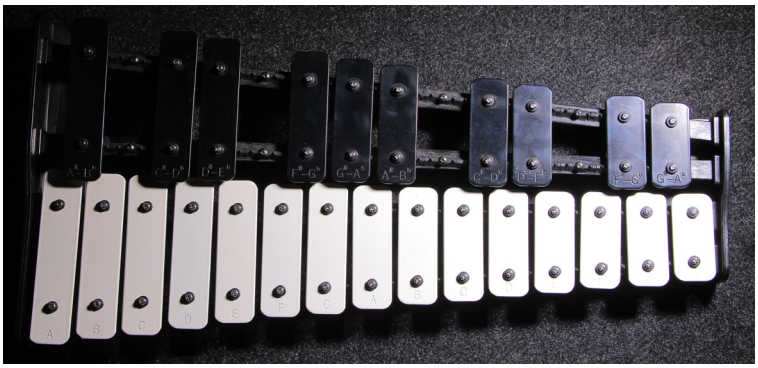
→ **Click the Instrument name selector to bring up a menu of the available instruments - and then select the desired instrument from the menu.**
Alternatively, click and drag up/down in the display above the selector to scroll through the instruments.

! **Depending on the instrument size (in MB), it could take a short moment before the entire instrument is loaded into RAM.**

! **Also note that the note ranges of the instruments extend outside their "natural" ranges, which could produce nice artificial effects.**

The available instruments are:

- **Alto Glockenspiel**



The Alto Glockenspiel was played with hard mallets and was recorded with close mics (stereo) in a large hall, slightly wet. The samples were taken from Soundiron's Alto Glockenspiel library.

- **Bamblong**



This instrument is also known as a bamboo log drum, from Indonesia. It was played with rubber mallets and was recorded with close mics (stereo) in a studio, dry. The samples were taken from Soundiron's Bamblong library.

- **Circle Bells Mallet**



This instruments is also known as Blossom Bells. It was played with rubber mallets and was recorded with close mics (stereo) in a studio, dry. The samples were taken from Soundiron's Circle Bells library.

- **Cylindrum**



This is an experimental instrument built from large diameter plastic piping, also known as a tubulum. It was played with rubber paddle mallets and was recorded with close mics (stereo) in a studio, dry. The samples were taken from Soundiron's Cylindrum library.

- **Imbibaphones**



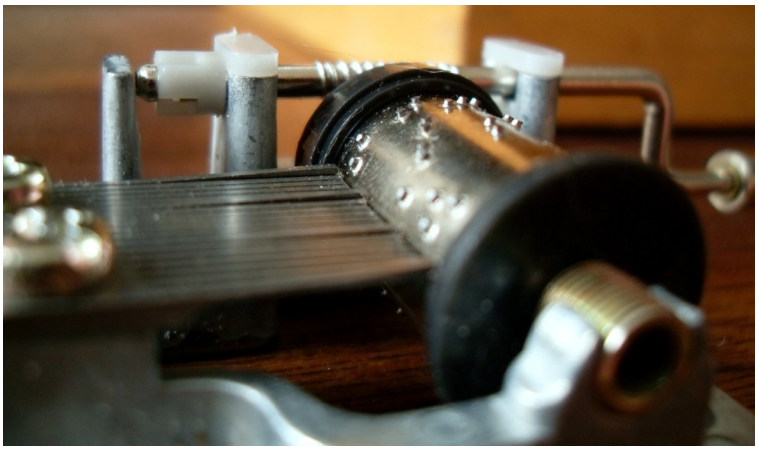
These are wine glasses played with rubber mallets. They were recorded with close mics (stereo) in a studio, dry. The samples were taken from Soundiron's "Imbibaphones" library.

- **Kalimba**



The Kalimba is also known as an mbira or thumb piano, from Africa. It was recorded with close mics (stereo) in a studio, dry. The samples were taken from Soundiron's Kalimba library.

- **Music Box**



This is a music box recorded with close mics (stereo) in a studio, dry. The samples were taken from Soundiron's Musique Box library.

- **Noah Bells**



The Noah Bells from India are played with the fingertips and were recorded with close mics (stereo) in a large hall, slightly wet. The samples were taken from Soundiron's Noah Bells library.

- **Steel Tones**



This instrument is also known as a hank drum or propane drum. It was played with felt mallets and was recorded with close mics (stereo) in a room, dry. The samples were taken from Soundiron's Steel Tones library.

- **Whale Drum**



The whale drum is a wooden box slit drum from Africa. It was played with rubber mallets and was recorded with close mics (stereo) in a studio, dry. The samples were taken from Soundiron's Whale Drum library.

S. Start (Sample Start)



→ **Turn the S.Start knob to set where in the sample the playback should start.**
Note that the effect could be different depending on the selected instrument.

Oct



→ **Set the octave transposition for the instrument.**
Range: 5 (+/-2) octaves.

! **Note that the note ranges of the instruments can be transposed far outside their "natural" ranges, which could produce nice artificial effects.**

Semi



→ **Set the pitch in semitone steps.**
Range: +/-12 semitones (two octaves).

Fine

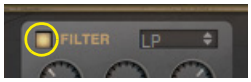


→ **Set the pitch in steps of 1 cent.**
Range: +/- 50 cents (down or up half a semitone).

The Filter section

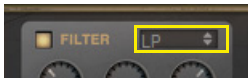


Filter On/Off



→ **Click the On/Off LED button to switch on/off the Filter section.**

Filter Type selector



→ **Click and drag up/down on the Filter Type selector to select one of the available filter types - or step through the filter types by clicking the Up/Down arrow buttons.**
The available filter types are:

- **LP**
 This is a lowpass filter with 12db/octave slope.
- **HP**
 This is a highpass filter with 12db/octave slope.
- **BP**
 This is a bandpass filter with 6db/octave slopes.
- **Comb**
 This is a comb filter for phaser/flanger type of effects.

Cutoff



→ **Set the cutoff/center frequency with the Cutoff knob.**
The cutoff parameter sets where in the frequency range you want the resonance and attenuation to appear.
Range: 20 Hz to 25 kHz.

Reso



→ **Set the resonance amount with the Reso knob.**
The resonance parameter amplifies the frequencies at, and around the cutoff/center frequency.

Env



→ **With the Env knob you set how much you want the Filter Envelope (see below) to affect the Cutoff frequency.**
Range: 0% to 100%.

Filter Envelope



The standard ADSR type envelope controls the filter cutoff frequency modulation over time. The ADSR envelope characteristics are described in detail in “Amp Envelope”.

Vel



→ **Turn the Vel knob to set how much the cutoff/center frequency should be modulated by Keyboard Velocity.**
Range: 0% to 100%.

Kbd



→ **Turn the KBD (Keyboard Track) knob to set how much the cutoff/center frequency should track incoming MIDI Notes.**
Range: 0% (no tracking (constant frequency)) to 100% (1 semitone per key).

The Amp section



Vel



➔ **Turn the Vel knob to set how much the amplitude should be modulated by Keyboard Velocity.**
Range: 0% to 100%.

Amp Envelope



The Amp Envelope is a standard ADSR envelope which controls the amplitude of the signal over time. The picture below shows the various stages of the ADSR envelope:



The ADSR envelope stages.

- **A(ttack)**
 When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the value set with the Master Volume knob. How long this should take, depends on the Attack setting. If the Attack is set to "0", the Volume value is reached instantly. If the Attack value is raised, it will take longer time before the Master Volume value is reached.

- **D(ecay)**
 After the Master Volume value has been reached, the level starts to drop. How long this should take is governed by the Decay parameter.

 If you want to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0", the Decay parameter should be set to a medium value and the Sustain level should be set to "0", so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you raise the Sustain parameter.
- **S(ustain)**
 The Sustain parameter determines the level the envelope should rest at, after the Decay stage. If you set Sustain to full level, the Decay setting is of no importance since the volume of the sound is never lowered.

 If you want to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

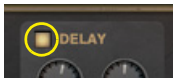
 But often a combination of Decay and Sustain is used to generate envelopes that rise up to the Volume value, then gradually decreases to finally land to rest on a level somewhere in-between zero and the Master Volume value. Note that Sustain represents a level, whereas the other envelope parameters represent times.
- **R(elease)**
 The Release parameter works just like the Decay parameter, except it determines the time it takes for the volume to drop back to zero after you release the key.

The Delay section



This is a stereo delay which affects all voices globally.

Delay On/Off



→ **Click the On/Off LED button to switch on/off the Delay section.**

Time



→ **Set the time between the delay repeats.**
If Sync is active (see below), the Time parameter controls the time divisions.

Feedback



→ **Set the number of delay repeats with the Feedback knob.**

Sync



→ **Click the Sync button to sync the delay time to the main sequencer Tempo.**
When active, the Time knob (see above) controls the time divisions.

Ping Pong



→ **Activate this to get the delay repeats alternating between the left and right channels.**
Note that this also doubles the delay tempo.

Damp



→ **Raise the Damp value to gradually cut off the high frequencies of the delay repeats.**

Amount



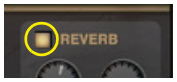
→ **Set the delay amount with the Amount knob.**
Note that the Delay effect is routed as a "send effect", with the Amount knob working as an "effect return" control. This means that if you play a short note with the Amount knob set to zero, you might still get a delay effect if you turn up the Amount afterwards (depending on the Feedback setting, see "Feedback").

The Reverb section



This is a stereo reverb which affects all voices globally.

Reverb On/Off



→ **Click the On/Off LED button to switch on/off the Reverb section.**

Time



→ **Set the reverberation duration time.**
In practice this sets the “size” of the reverberation “chamber/room/hall”.

Pre-Delay



→ **Set the pre-delay time of the reverb.**
Range: 0-200 ms.

Hi Damp



→ **Raise the Hi Damp value to cut off the high frequencies of the reverb and thereby create a smoother, warmer effect.**

Lo Damp



→ **Raise the Lo Damp value to cut off the low frequencies of the reverb signal, to make the reverb effect less "muddy".**

Amount



→ **Set the reverb amount with the Amount knob.**
Note that the Reverb effect is routed as a "send effect", with the Amount knob working as an "effect return" control. This means that if you play a short note with the Amount knob set to zero, you might still get a reverb effect if you turn up the Amount afterwards (depending on the Time setting, see "Time").

Connections



! **Remember that CV connections are NOT stored in the Klang patches! If you want to store CV connections between devices, put them in a Combinator device and save the Combi patch.**

Sequencer Control inputs

The Sequencer Control CV and Gate inputs allow you to play Klang from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity. There are also inputs for modulating the Pitch Bend and Mod Wheel parameters.

Modulation Inputs

These control voltage (CV) inputs can be used for modulating the Filter Cutoff and Resonance parameters, as well as the Master Volume level.

Audio Out

These are the main audio outputs. When you create a new Klang device, these outputs are auto-routed to the first available outputs in the I/O device.

Chapter 24
Pangea
World Instruments

Introduction



Pangea World Instruments features a unique assortment of rare instruments from all over the world - perfect for spicing up any music style. Each of the multi-sampled instruments can also be tailored and processed in the high-quality filter, amp, delay and reverb sections.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Panel overview

The Pangea front panel contains the following sections:

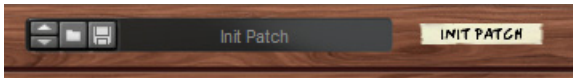


The Pangea front panel sections.

- **1. Patch Selector (for browsing, loading and saving patches).**
- **2. Master Volume.**
- **3. Global performance and "play" controls.**
- **4. Instruments section.**
- **5. Filter section.**
- **6. Amp Envelope section.**
- **7. Delay effect section.**
- **8. Reverb effect section.**

Using Pangea

Loading and saving patches



Loading and saving patches is done in the same way as with any other internal Reason device, see "Loading patches" and "Saving patches" for details.

Global performance and "play" controls



Note

The Note LED lights up each time Pangea receives a MIDI Note On.

Range

→ **Set the desired Pitch Bend range for the "Pitch" wheel with the up/down buttons, or by click-holding on the display and dragging up/down.**
Range: +/-12 semitones (+/-1 octave) in steps of +/-1 semitone.

Pitch

The Pitch bend wheel can be used for bending note pitches up and down. Pangea also responds to Pitch Bend MIDI data from a connected MIDI master keyboard. You set the desired Pitch bend Range with the "Range" control to the right of the Mod wheel.

Mod

The Mod wheel can be used for controlling the three predefined parameters to the right of the Mod wheel:

- **S. Start**
 Here you set how/if the Mod wheel should affect the Sample Start position of the currently selected instrument. The parameter is bi-polar, with zero modulation at the 12 o'clock position. A negative value moves the sample start back and a positive value moves it forward.

! **Note that if the S.Start parameter in the Instruments section (see "S. Start (Sample Start)") is at 0 ms, the sample start cannot be moved back any further. Similarly, if the S.Start parameter in the Instruments section is at 150 ms, the sample start cannot be moved forward any further.**

- **F. Freq**
 Here you set how/if the Mod wheel should affect the Filter Cutoff parameter. The parameter is bi-polar, with zero modulation at the 12 o'clock position.

! **Note that if the Cutoff parameter in the Filter section (see "Cutoff") is at 20 Hz, the frequency cannot be lowered any further. Similarly, if the Cutoff parameter in the Filter section is at 25 kHz, the frequency cannot be raised any further.**

- **Level**
 Here you set how/if the Mod wheel should affect the Master Volume parameter. The parameter is bi-polar, with zero modulation at the 12 o'clock position.

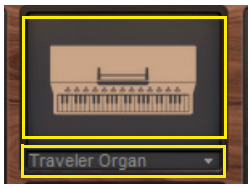
! **The volume can never be modulated louder than the maximum +12 dB limit of the Master Volume control.**

Panel controls

The Instruments section



Instrument selector



→ **Click the Instrument name selector to bring up a menu of the available instruments - and then select the desired instrument from the menu.**
Alternatively, click and drag up/down in the display above the selector to scroll through the instruments.

! **Depending on the instrument size (in MB), it could take a short moment before the entire instrument is loaded into RAM.**

! **Also note that the note ranges of the instruments extend outside their "natural" ranges, which could produce nice artificial effects.**

The available instruments are:

- **Acoustic Saz**



This is a 5-string electro-acoustic hybrid saz baglema from Turkey, also known as Turkish guitar. It was played with a hard pick and the strings were recorded with external close mics (stereo) in a studio, dry. The samples are from Soundiron's Acoustic Saz library.

- **Angklung**



This is an18-piece tuned bamboo rattle instrument from Indonesia. It was recorded with close mics (stereo) in a large hall. The samples are from Soundiron's Angklung library.

- **Bizarre Sitar**



This is a small 8-string sitar from India. It was played with a hard pick and was recorded with close mics (stereo) in a studio, dry. The samples are from Soundiron's Bizarre Sitar library.

- **Harp Guitar**



This is a custom instrument designed by Brad Hoyt. It was played with a hard pick and was recorded with close mics (stereo) in a studio, dry. The samples are from Soundiron's Brad Hoyt Harp Guitar library.

- **Kinderklavier**



This is a children's toy steel tine piano from Germany. It was recorded with close mics (stereo) in a studio, dry. The samples are from Soundiron's Kinderklavier library.

- **Lakeside Pipe Organ**



This is a large church pipe organ recorded with close mics (stereo) in a large hall, wet. The samples are from Soundiron's Lakeside Pipe Organ library.

- **Little Wooden Flutes**



This is a Native American walnut 6-hole flute. It was recorded with close mics (stereo) in a studio, dry. The samples are from Soundiron's Little Wooden Flutes library.

- **Little Pump Reeds**



This is a pumped reed instrument related to a harmonium, from India. It was recorded with close mics (stereo) in a studio, dry. The samples are from Soundiron's Little Pump Reeds library.

- **Struck Grand Piano**



This is a grand piano, with the strings being struck with a small metal hammer. It was recorded with close mics (stereo) in a large hall. The samples are from Soundiron's Struck Grand Piano library.

- **Traveler Organ**



This is a mechanically operated antique organ, also known as a traveling organ. It is operated by pumping in air using the two pedals and then playing the keyboard. It was recorded with close mics (stereo) in a studio, dry. The samples are from Soundiron's Traveler Organ library.

- **Zitherette**



This is an 8 string fretless zither played with a hard pick. It was recorded with a close mic (mono) in a studio, dry. The samples are from Soundiron's Zitherette library.

S. Start (Sample Start)



→ **Turn the S.Start knob to set where in the sample the playback should start.**
Note that the effect could be different depending on the selected instrument.

Oct



→ **Set the octave transposition for the instrument.**
Range: 5 (+/-2) octaves.

! **Note that the note ranges of the instruments can be transposed far outside their “natural” ranges, which could produce nice artificial effects.**

Semi



→ **Set the pitch in semitone steps.**
Range: +/-12 semitones (two octaves).

Fine

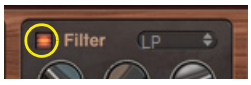


→ **Set the pitch in steps of 1 cent.**
Range: +/- 50 cents (down or up half a semitone).

The Filter section

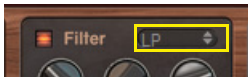


Filter On/Off



→ **Click the On/Off LED button to switch on/off the Filter section.**

Filter Type selector



→ **Click and drag up/down on the Filter Type selector to select one of the available filter types - or step through the filter types by clicking the Up/Down arrow buttons.**
The available filter types are:

- **LP**
 This is a lowpass filter with 12db/octave slope.
- **HP**
 This is a highpass filter with 12db/octave slope.
- **BP**
 This is a bandpass filter with 6db/octave slopes.
- **Comb**
 This is a comb filter for phaser/flanger type of effects.

Cutoff



→ **Set the cutoff/center frequency with the Cutoff knob.**
The cutoff parameter sets where in the frequency range you want the resonance and attenuation to appear.
Range: 20 Hz to 25 kHz.

Reso



→ **Set the resonance amount with the Reso knob.**
The resonance parameter amplifies the frequencies at, and around the cutoff/center frequency.

Env



→ **With the Env knob you set how much you want the Filter Envelope (see below) to affect the Cutoff frequency.**
Range: 0% to 100%.

Filter Envelope



The standard ADSR type envelope controls the filter cutoff frequency modulation over time. The ADSR envelope characteristics are described in detail in “Amp Envelope”.

Vel



→ **Turn the Vel knob to set how much the cutoff/center frequency should be modulated by Keyboard Velocity.**
Range: 0% to 100%.

Kbd



→ **Turn the KBD (Keyboard Track) knob to set how much the cutoff/center frequency should track incoming MIDI Notes.**
Range: 0% (no tracking (constant frequency)) to 100% (1 semitone per key).

The Amp section

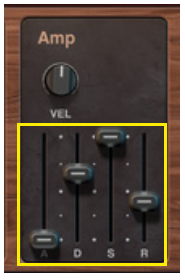


Vel



→ **Turn the Vel knob to set how much the amplitude should be modulated by Keyboard Velocity.**
Range: 0% to 100%.

Amp Envelope



The Amp Envelope is a standard ADSR envelope which controls the amplitude of the signal over time. The picture below shows the various stages of the ADSR envelope:



The ADSR envelope stages.

- **A(ttack)**
 When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the value set with the Master Volume knob. How long this should take, depends on the Attack setting. If the Attack is set to "0", the Volume value is reached instantly. If the Attack value is raised, it will take longer time before the Master Volume value is reached.
- **D(ecay)**
 After the Master Volume value has been reached, the level starts to drop. How long this should take is governed by the Decay parameter.

 If you want to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0", the Decay parameter should be set to a medium value and the Sustain level should be set to "0", so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you raise the Sustain parameter.
- **S(ustain)**
 The Sustain parameter determines the level the envelope should rest at, after the Decay stage. If you set Sustain to full level, the Decay setting is of no importance since the volume of the sound is never lowered.

 If you want to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

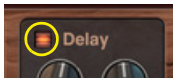
 But often a combination of Decay and Sustain is used to generate envelopes that rise up to the Volume value, then gradually decreases to finally land to rest on a level somewhere in-between zero and the Master Volume value. Note that Sustain represents a level, whereas the other envelope parameters represent times.
- **R(elease)**
 The Release parameter works just like the Decay parameter, except it determines the time it takes for the volume to drop back to zero after you release the key.

The Delay section



This is a stereo delay which affects all voices globally.

Delay On/Off



→ **Click the On/Off LED button to switch on/off the Delay section.**

Time



→ **Set the time between the delay repeats.**
If Sync is active (see below), the Time parameter controls the time divisions.

Feedback



→ **Set the number of delay repeats with the Feedback knob.**

Sync



→ **Click the Sync button to sync the delay time to the main sequencer Tempo.**
When active, the Time knob (see above) controls the time divisions.

Ping Pong



→ **Activate this to get the delay repeats alternating between the left and right channels.**
Note that this also doubles the delay tempo.

Damp



→ **Raise the Damp value to gradually cut off the high frequencies of the delay repeats.**

Amount



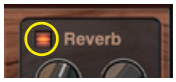
→ **Set the delay amount with the Amount knob.**
Note that the Delay effect is routed as a "send effect", with the Amount knob working as an "effect return" control. This means that if you play a short note with the Amount knob set to zero, you might still get a delay effect if you turn up the Amount afterwards (depending on the Feedback setting, see "Feedback").

The Reverb section



This is a stereo reverb which affects all voices globally.

Reverb On/Off



→ **Click the On/Off LED button to switch on/off the Reverb section.**

Time



→ **Set the reverberation duration time.**
In practice this sets the “size” of the reverberation “chamber/room/hall”.

Pre-Delay



→ **Set the pre-delay time of the reverb.**
Range: 0-200 ms.

Hi Damp



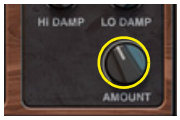
→ **Raise the Hi Damp value to cut off the high frequencies of the reverb and thereby create a smoother, warmer effect.**

Lo Damp



→ **Raise the Lo Damp value to cut off the low frequencies of the reverb signal, to make the reverb effect less “muddy”.**

Amount



→ **Set the reverb amount with the Amount knob.**
Note that the Reverb effect is routed as a “send effect”, with the Amount knob working as an “effect return” control. This means that if you play a short note with the Amount knob set to zero, you might still get a reverb effect if you turn up the Amount afterwards (depending on the Time setting, see “Time”).

Connections



! **Remember that CV connections are NOT stored in the Pangea patches! If you want to store CV connections between devices, put them in a Combinator device and save the Combi patch.**

Sequencer Control inputs

The Sequencer Control CV and Gate inputs allow you to play Pangea from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity. There are also inputs for modulating the Pitch Bend and Mod Wheel parameters.

Modulation Inputs

These control voltage (CV) inputs can be used for modulating the Filter Cutoff and Resonance parameters, as well as the Master Volume level.

Audio Out

These are the main audio outputs. When you create a new Pangea device, these outputs are auto-routed to the first available outputs in the I/O device.

Chapter 25
Humana
Vocal Ensemble

Introduction



Humana Vocal Ensemble features a great selection of male and female vocal samples - perfect for any music style. The multi-sampled vocal sound sets can also be tailored and processed in the high-quality filter, amp, delay and reverb sections.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Panel overview

The Humana front panel contains the following sections:

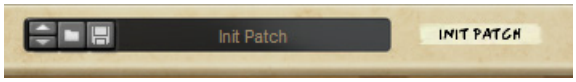


The Humana front panel sections.

- **1. Patch Selector (for browsing, loading and saving patches).**
- **2. Master Volume.**
- **3. Global performance and "play" controls.**
- **4. Instruments section.**
- **5. Filter section.**
- **6. Amp Envelope section.**
- **7. Delay effect section.**
- **8. Reverb effect section.**

Using Humana

Loading and saving patches



Loading and saving patches is done in the same way as with any other internal Reason device, see "Loading patches" and "Saving patches" for details.

Global performance and "play" controls



Note

The Note LED lights up each time Humana receives a MIDI Note On.

Range

→ **Set the desired Pitch Bend range for the "Pitch" wheel with the up/down buttons, or by click-holding on the display and dragging up/down.**
Range: +/-12 semitones (+/-1 octave) in steps of +/-1 semitone.

Pitch

The Pitch bend wheel can be used for bending note pitches up and down. Humana also responds to Pitch Bend MIDI data from a connected MIDI master keyboard. You set the desired Pitch bend Range with the "Range" control to the right of the Mod wheel.

Mod

The Mod wheel can be used for controlling the three predefined parameters to the right of the Mod wheel:

- **S. Start**
 Here you set how/if the Mod wheel should affect the Sample Start position of the currently selected instrument. The parameter is bi-polar, with zero modulation at the 12 o'clock position. A negative value moves the sample start back and a positive value moves it forward.

! **Note that if the S.Start parameter in the Instruments section (see "S. Start (Sample Start)") is at 0 ms, the sample start cannot be moved back any further. Similarly, if the S.Start parameter in the Instruments section is at 150 ms, the sample start cannot be moved forward any further.**

- **F. Freq**
 Here you set how/if the Mod wheel should affect the Filter Cutoff parameter. The parameter is bi-polar, with zero modulation at the 12 o'clock position.

! **Note that if the Cutoff parameter in the Filter section (see "Cutoff") is at 20 Hz, the frequency cannot be lowered any further. Similarly, if the Cutoff parameter in the Filter section is at 25 kHz, the frequency cannot be raised any further.**

- **Level**
 Here you set how/if the Mod wheel should affect the Master Volume parameter. The parameter is bi-polar, with zero modulation at the 12 o'clock position.

! **The volume can never be modulated louder than the maximum +12 dB limit of the Master Volume control.**

Panel controls

The Instruments section



Instrument selector



→ **Click the Instrument name selector to bring up a menu of the available instruments - and then select the desired instrument from the menu.**
Alternatively, click and drag up/down in the display above the selector to scroll through the instruments.

! **Depending on the instrument size (in MB), it could take a short moment before the entire instrument is loaded into RAM.**

! **Also note that the note ranges of the instruments extend outside their "natural" ranges, which could produce nice artificial effects.**

The available instruments are:

- **Mars ah**



A male vocal ensemble singing sustained “Ah’s” (forte). Recorded with stage mics (stereo) in a large hall, wet. Conducted by Robert Geary. The samples are from Soundiron's Olympus Symphonic Choir.

- **Mars oo**
 A male vocal ensemble singing sustained “Oo’s” (piano). Recorded with stage mics (stereo) in a large hall, wet. Conducted by Robert Geary. The samples are from Soundiron's Olympus Symphonic Choir.
- **Mars ah Staccato**
 A male vocal ensemble singing staccato “Ah’s” (forte). Recorded with stage mics (stereo) in a large hall, wet. Conducted by Robert Geary. The samples are from Soundiron's Olympus Symphonic Choir.
- **Mars oo Staccato**
 A male vocal ensemble singing staccato “Oo’s” (piano). Recorded with stage mics (stereo) in a large hall, wet. Conducted by Robert Geary. The samples are from Soundiron's Olympus Symphonic Choir.
- **Venus ah**



A female vocal ensemble singing sustained “Ah’s” (forte). Recorded with stage mics (stereo) in a large hall, wet. Conducted by Robert Geary. The samples are from Soundiron's Olympus Symphonic Choir.

- **Venus oo**
 A female vocal ensemble singing sustained “Oo’s” (piano). Recorded with stage mics (stereo) in a large hall, wet. Conducted by Robert Geary. The samples are from Soundiron's Olympus Symphonic Choir.
- **Venus ah Staccato**
 A female vocal ensemble singing staccato “Ah’s” (forte). Recorded with stage mics (stereo) in a large hall, wet. Conducted by Robert Geary. The samples are from Soundiron's Olympus Symphonic Choir.
- **Venus oo Staccato**
 A female vocal ensemble singing staccato “Oo’s” (piano). Recorded with stage mics (stereo) in a large hall, wet. Conducted by Robert Geary. The samples are from Soundiron's Olympus Symphonic Choir.

- **Mercury ah**



A boys' choir singing sustained "Ah's" (forte). Recorded with stage mics (stereo) in a large hall, wet. Conducted by Robert Geary. The samples are from Soundiron's Mercury Symphonic Boys' Choir.

- **Female Soprano ah**



Female soprano Nichole Dechaine singing sustained "Ah's" (forte). Recorded with close mic (mono) in a studio, dry. The samples are from Soundiron's Voices Of Rapture.

- **Female Alto ah**



Female alto Kindra Scharich singing sustained “Ah’s” (forte). Recorded with close mic (mono) in a studio, dry. The samples are from Soundiron's Voices Of Rapture.

- **Male Tenor ah**



Male tenor Brian Thorsett singing sustained “Ah’s” (forte). Recorded with close mic (mono) in a studio, dry. The samples are from Soundiron's Voices Of Rapture.

- **Male Bass ah**



Male bass Joseph Trumbo singing sustained “Ah’s” (forte). Recorded with close mic (mono) in a studio, dry. The samples are from Soundiron's Voices Of Rapture.

- **Female ah**



Female alto soloist Francesca Genco singing sustained “Ah’s” (forte). Recorded with close mic (mono) in a studio, dry. The samples are from Soundiron's Voices Of Gaia.

- **Female ah 2**



Female alto soloist Linda Strawberry singing sustained “Ah’s” (forte). Recorded with close mic (mono) in a studio, dry. The samples are from Soundiron's Voices Of Gaia.

- **Male ah**



Male tenor soloist Brian Lane singing sustained “Ah’s” (forte). Recorded with close mic (mono) in a studio, dry. The samples are from Soundiron's Voices Of Gaia.

S. Start (Sample Start)



→ **Turn the S.Start knob to set where in the sample the playback should start.**
Note that the effect could be different depending on the selected instrument.

Oct



→ **Set the octave transposition for the instrument.**
Range: 5 (+/-2) octaves.

! **Note that the note ranges of the instrumentsan be transposed far outside their "natural" ranges, which could produce nice artificial effects.**

Semi



→ **Set the pitch in semitone steps.**
Range: +/-12 semitones (two octaves).

Fine



→ **Set the pitch in steps of 1 cent.**
Range: +/- 50 cents (down or up half a semitone).

The Filter section



Filter On/Off



→ **Click the On/Off LED button to switch on/off the Filter section.**

Filter Type selector



→ **Click and drag up/down on the Filter Type selector to select one of the available filter types - or step through the filter types by clicking the Up/Down arrow buttons.**
The available filter types are:

- **LP**
 This is a lowpass filter with 12db/octave slope.
- **HP**
 This is a highpass filter with 12db/octave slope.
- **BP**
 This is a bandpass filter with 6db/octave slopes.
- **Comb**
 This is a comb filter for phaser/flanger type of effects.

Cutoff



→ **Set the cutoff/center frequency with the Cutoff knob.**
The cutoff parameter sets where in the frequency range you want the resonance and attenuation to appear.
Range: 20 Hz to 25 kHz.

Reso



→ **Set the resonance amount with the Reso knob.**
The resonance parameter amplifies the frequencies at, and around the cutoff/center frequency.

Env



→ **With the Env knob you set how much you want the Filter Envelope (see below) to affect the Cutoff frequency.**
Range: 0% to 100%.

Filter Envelope



The standard ADSR type envelope controls the filter cutoff frequency modulation over time. The ADSR envelope characteristics are described in detail in “Amp Envelope”.

Vel



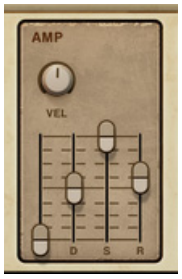
- **Turn the Vel knob to set how much the cutoff/center frequency should be modulated by Keyboard Velocity.**
 Range: 0% to 100%.

Kbd



- **Turn the KBD (Keyboard Track) knob to set how much the cutoff/center frequency should track incoming MIDI Notes.**
 Range: 0% (no tracking (constant frequency)) to 100% (1 semitone per key).

The Amp section

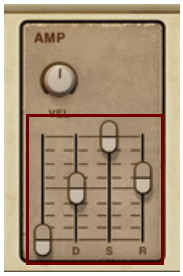


Vel



- **Turn the Vel knob to set how much the amplitude should be modulated by Keyboard Velocity.**
 Range: 0% to 100%.

Amp Envelope



The Amp Envelope is a standard ADSR envelope which controls the amplitude of the signal over time. The picture below shows the various stages of the ADSR envelope:



The ADSR envelope stages.

- **A(ttack)**
 When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the value set with the Master Volume knob. How long this should take, depends on the Attack setting. If the Attack is set to "0", the Volume value is reached instantly. If the Attack value is raised, it will take longer time before the Master Volume value is reached.
- **D(ecay)**
 After the Master Volume value has been reached, the level starts to drop. How long this should take is governed by the Decay parameter.

 If you want to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0", the Decay parameter should be set to a medium value and the Sustain level should be set to "0", so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you raise the Sustain parameter.
- **S(ustain)**
 The Sustain parameter determines the level the envelope should rest at, after the Decay stage. If you set Sustain to full level, the Decay setting is of no importance since the volume of the sound is never lowered.

 If you want to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

 But often a combination of Decay and Sustain is used to generate envelopes that rise up to the Volume value, then gradually decreases to finally land to rest on a level somewhere in-between zero and the Master Volume value. Note that Sustain represents a level, whereas the other envelope parameters represent times.

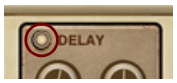
- **R(elease)**
 The Release parameter works just like the Decay parameter, except it determines the time it takes for the volume to drop back to zero after you release the key.

The Delay section



This is a stereo delay which affects all voices globally.

Delay On/Off



→ **Click the On/Off LED button to switch on/off the Delay section.**

Time



→ **Set the time between the delay repeats.**
If Sync is active (see below), the Time parameter controls the time divisions.

Feedback



→ **Set the number of delay repeats with the Feedback knob.**

Sync



→ **Click the Sync button to sync the delay time to the main sequencer Tempo.**
When active, the Time knob (see above) controls the time divisions.

Ping Pong



→ **Activate this to get the delay repeats alternating between the left and right channels.**
Note that this also doubles the delay tempo.

Damp



→ **Raise the Damp value to gradually cut off the high frequencies of the delay repeats.**

Amount



→ **Set the delay amount with the Amount knob.**
Note that the Delay effect is routed as a "send effect", with the Amount knob working as an "effect return" control. This means that if you play a short note with the Amount knob set to zero, you might still get a delay effect if you turn up the Amount afterwards (depending on the Feedback setting, see "Feedback").

The Reverb section



This is a stereo reverb which affects all voices globally.

Reverb On/Off



→ **Click the On/Off LED button to switch on/off the Reverb section.**

Time



→ **Set the reverberation duration time.**
In practice this sets the “size” of the reverberation “chamber/room/hall”.

Pre-Delay



→ **Set the pre-delay time of the reverb.**
Range: 0-200 ms.

Hi Damp



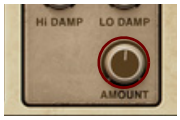
→ **Raise the Hi Damp value to cut off the high frequencies of the reverb and thereby create a smoother, warmer effect.**

Lo Damp



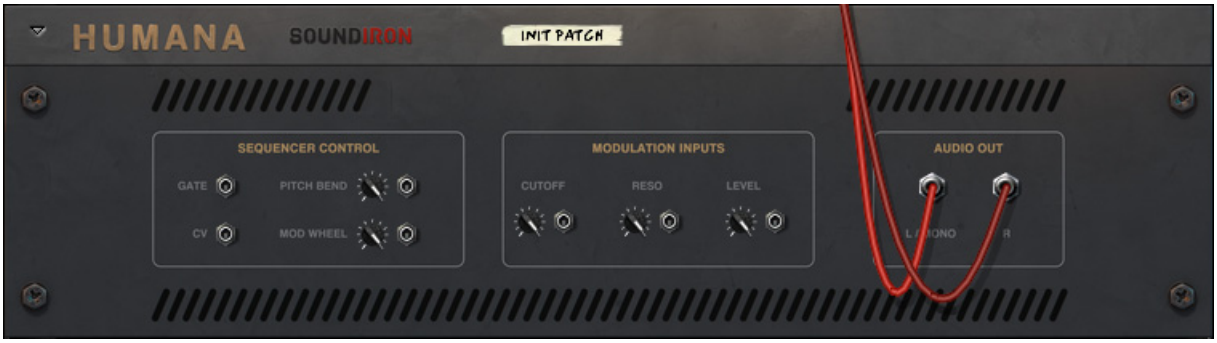
→ **Raise the Lo Damp value to cut off the low frequencies of the reverb signal, to make the reverb effect less “muddy”.**

Amount



→ **Set the reverb amount with the Amount knob.**
Note that the Reverb effect is routed as a “send effect”, with the Amount knob working as an “effect return” control. This means that if you play a short note with the Amount knob set to zero, you might still get a reverb effect if you turn up the Amount afterwards (depending on the Time setting, see “Time”).

Connections



! **Remember that CV connections are NOT stored in the Humana patches! If you want to store CV connections between devices, put them in a Combinator device and save the Combi patch.**

Sequencer Control inputs

The Sequencer Control CV and Gate inputs allow you to play Humana from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity. There are also inputs for modulating the Pitch Bend and Mod Wheel parameters.

Modulation Inputs

These control voltage (CV) inputs can be used for modulating the Filter Cutoff and Resonance parameters, as well as the Master Volume level.

Audio Out

These are the main audio outputs. When you create a new Humana device, these outputs are auto-routed to the first available outputs in the I/O device.

Chapter 26
NN-XT Sampler

Introduction

The basic functions of the NN-XT are very similar to those of its sampler companion in the Reason rack - the NN-19 (see "NN-19 Sampler"). Just like the NN-19, NN-XT lets you load samples and create multi-sample patches by mapping samples across the keyboard. The sound can then be modified by a comprehensive set of synth-type parameters. There are however some major differences between the two. The NN-XT has:

- **Support for SoundFonts.**
 Presets and samples from SoundFont banks can be loaded and used in the NN-XT (see "Loading SoundFonts").
- **8 stereo output pairs.**
 This makes it possible to route different samples to different mixer channels for individual effect processing (see "Out").
- **The possibility to create layered sounds.**
 This is done by mapping several samples across the same keyboard range (see "Creating layered sounds").
- **The possibility to create sounds that only play over certain velocity ranges, velocity switched key maps and velocity crossfading.**
 See "Setting velocity range for a Zone".
- **Key maps with individual synth parameter settings for each sample.**
 See "Synth parameters".

Even though the NN-XT is a more advanced sample player than NN-19, it should not be considered as a successor to the NN-19, but rather as a complement to it. The NN-19 will for example probably still be the sampler of choice for those of you who want to be able to quickly load a couple of samples and start playing, since that particular aspect takes a little more doing with the NN-XT.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

Panel overview

The main panel

When the NN-XT is added to the rack, you will initially only see the main panel.



The NN-XT main panel.

The main panel is where you load complete sample patches. It also contains the “global controls”. These are controls that affect and modify the sound of entire patches rather than the individual key zones.

The Remote Editor panel

To show/hide the remote editor panel, use the fold/unfold arrow at the bottom left.



The remote editor panel is where you load individual samples, create key maps, modify the sound of the samples with synth parameters etc.

! **The main panel of the NN-XT can be folded like any other Reason device. Note that folding the main panel will also fold the remote editor regardless of its current state.**

Loading complete Patches and REX files

As previously alluded, you can load complete sample patches as well as individual samples into the NN-XT.

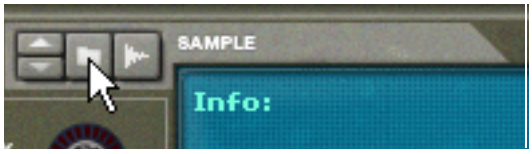
- **A patch is a complete "sound package". It contains information about all the samples used, assigned key zones, associated panel settings etc. Loading a sample patch is done by using the patch browser on the main panel, and works in the same way as with any other Reason device.**



The Browse Patch button on the main panel.

For general instructions on how to load and save patches, please see "Loading patches" and "Saving patches".

- **Loading separate samples is done in a similar way, but via the sample browser on the remote editor panel. If you load samples, map them across keyboard ranges and set up the sound the way you want it, you can save your settings as a Patch for easy access later.**



The Browse Sample button on the remote editor.

More about loading samples later in this chapter.

Loading NN-XT Patches

NN-XT Patches are patches made specifically for the NN-XT. Reason ships with a large number of NN-XT Patches, some in the Factory Sound Bank but most in the Orkester Sound Bank. NN-XT Patches have the extension ".sxt".

1. **Click the Browse Patch button on the front panel to set browse focus to the NN-XT device.**
2. **Navigate to the folder that contains the NN-XT patch you wish to load, select it and click Load in the Browser.**

→ **Alternatively, drag an NN-XT patch from the Browser and drop it on the NN-XT device in the rack.**
The panel is dimmed in orange and the Patch Replace symbol appears in the center.

Loading NN-19 Patches

NN-19 Patches have the extension ".smp". Note that when loading NN-19 patches into the NN-XT, some parameters will not be applicable since the NN-19 and the NN-XT to some extent differ from each other in terms of controls. In these cases, the concerned parameters will either be ignored by the NN-XT or mapped to the most equivalent control.

1. **Click the Browse Patch button on the front panel to set browse focus to the NN-XT device.**
2. **Navigate to the folder that contains the NN-19 patch you wish to load, select it and click Load in the Browser.**

→ **Alternatively, drag an NN-19 patch from the Browser and drop it on the NN-XT device in the rack.**
The panel is dimmed in orange and the Patch Replace symbol appears in the center.

Loading SoundFonts

The SoundFont format was developed by E-mu systems in collaboration with Creative Technologies. It is a standardized data format containing wavetable synthesized audio and information on how it should be played back in wavetable synthesizers - typically on audio cards. The SoundFont format is an open standard so there is a vast amount of SoundFont banks and SoundFont compatible banks developed by third parties.

Loading SoundFonts is no different from loading NN-XT Patches. As with NN-19 Patches, the NN-XT does its best to map all the SoundFont settings to NN-XT parameters.

You can load SoundFont presets by using the patch browser, and single SoundFont samples by using the sample browser.

Loading complete REX files as Patches

REX files are music loops created in the ReCycle program or when editing audio clips inline in Reason (see "Bounce Clip to REX Loop"). In Reason, REX files are primarily used in the Dr. Octo Rex loop player, but they can be used in the NN-XT as well. Possible extensions are ".rx2", ".rcy" and ".rex".

1. **Click the Browse Patch button on the front panel to set browse focus to the NN-XT device.**
2. **Navigate to the folder that contains the REX loop you wish to load, select it and click Load in the Browser.**

→ **Alternatively, drag a REX loop from the Browser and drop it on the NN-XT device in the rack.**
The panel is dimmed in orange and the Patch Replace symbol appears in the center.

When loading a REX file, each slice in the file is assigned to one key, chromatically, starting from C1. All parameters are set to their default settings.

When using REX files in the Dr. Octo Rex loop player, it is possible to make a track play the slices in order to recreate the original loop. To do the same in the NN-XT requires a few extra steps.

1. **Use the Browser to load the REX file into an NN-XT sampler.**
2. **Create a Dr. Octo Rex loop player and load the same REX file into a Loop Slot of this device.**
3. **Use the Copy Loop To Track feature on the Dr. Octo Rex to create playback data (a group) on the track assigned to the Dr. Octo Rex.**
4. **Move that group to the track that plays the NN-XT and play it back from there.**
5. **Delete the Dr. Octo Rex loop player.**

Using the main panel

All of the controls on the main panel are used for globally modifying certain parameters for *all* of the samples in a patch, by the same amount.

! **Movements of the parameters on the main panel can be recorded as automation. However, controls on the remote editor panel (described later) can not!**

The following is a description of the controls and parameters on the main panel.

The Pitch and Modulation wheels



Most MIDI keyboards come equipped with Pitch Bend and Modulation wheels. The NN-XT features settings for how incoming MIDI Pitch Bend and Modulation wheel messages should affect the sound. The wheels on the NN-XT will also mirror the movements of the wheels on your MIDI keyboard.

If you don't have Pitch Bend or Modulation controls on your keyboard, or if you aren't using a keyboard at all, you can use the two fully functional wheels on the NN-XT to apply real time modulation and pitch bend.

- **The Pitch Bend wheel is used for "bending" the played notes up and down to change their pitch - much like bending the strings on a guitar or other string instrument. The Pitch Bend Range is set on the remote editor panel (see "Pitch Bend Range").**
- **The Modulation wheel can be used for applying modulation to the sound while you're playing. It can also be used for controlling a number of other parameters, as described in "The Modulation controls".**

The External Control wheel



This section can be used in three ways:

Receiving MIDI controller messages from external sources

NN-XT can receive common MIDI controller messages, and route these to various parameters. You use the "Source" selector switch to determine which type of message should be received:

- **Aftertouch (Channel Pressure)**

- **Expression Pedal**
- **Breath Control**

If your MIDI keyboard is capable of sending aftertouch messages, and/or if you have connected an expression pedal or a breath controller to it, you can use these to modulate NN-XT parameters. Which parameters should be modulated is set in the remote editor panel (see "The Modulation controls").

Recording MIDI controller messages with the wheel

The wheel in the external control section can be used for recording any or all of the three MIDI controller message types into the main sequencer. If your MIDI keyboard isn't capable of sending aftertouch messages or you don't have access to an expression pedal or a breath controller, you can use the wheel instead. This is done just as with any other automation recording.

High Quality Interpolation



This switch turns High Quality Interpolation on and off. When it is activated, the sample pitch is calculated using a more advanced interpolation algorithm. This results in better audio quality, especially for samples with a lot of high frequency content.

- **High Quality Interpolation uses more computer power - so if you don't need it, it's a good idea to turn it off! Listen to the sounds in a context and determine whether you think this setting makes any difference.**

Global Controls



All of these knobs change the values of various parameters in the remote editor panel and affect *all* loaded samples. Thus they can be used for quickly adjusting the overall sound.

The knobs are bi-polar, which means that when they are centered, no parameter change is applied. By turning them to the right you increase the corresponding value, and by turning them to the left, you decrease the value.

Again, the movements of these parameters can be recorded as automation. This is done just as with any other automation recording.

The controls are, from left to right:

Filter

These two knobs each control a parameter of the filter (see "The Filter section"). Note that the filter must be on for these to have any effect.

- **Frequency**
 This changes the cutoff frequency of the filter.
- **Resonance**
 This changes the resonance parameter of the filter, meaning - the filter characteristic, or quality.

Amp Envelope

These three knobs control the Amplitude Envelope (see "The Amplitude Envelope") in the following way:

- **Attack**
 This changes the Attack value of the Amplitude Envelope. That is, how long it should take for the sound to reach full level after you press a key on your keyboard.
- **Decay**
 This changes the Decay value of the Amplitude Envelope. Decay determines how long it should take for the sound to go back to the sustain level after it has reached full value (see "The Amplitude Envelope") and the key that triggered the sound is still being pressed.
- **Release**
 This changes the Release value of the Amplitude Envelope. Release works just like Decay with the exception that it determines how long it should take for the sound to become silent after the key has been *released*.

Mod Envelope

This knob controls the Decay value of the Modulation Envelope (see "The Modulation Envelope"). Also see above for a brief description of Decay.

Master Volume

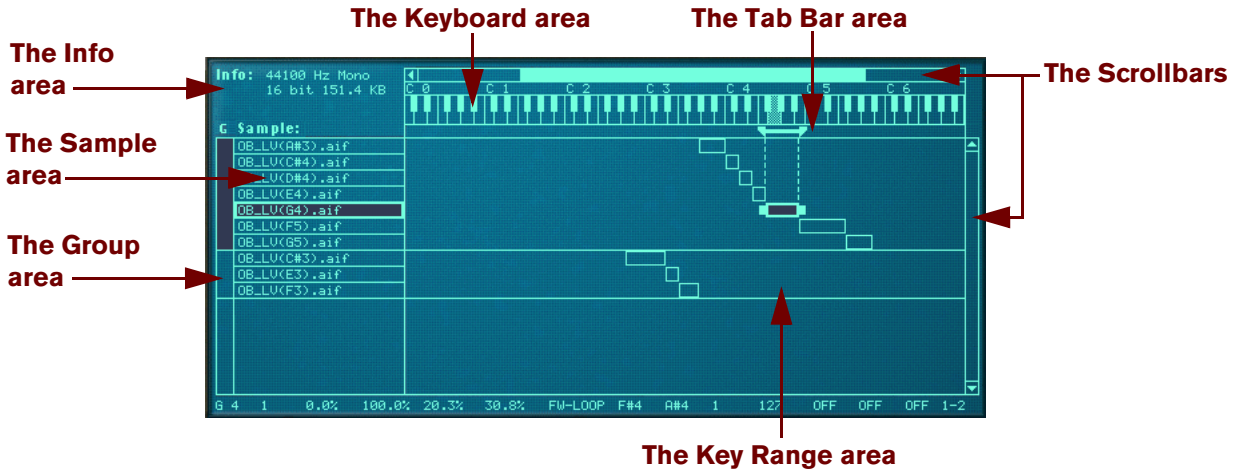
This controls the main volume out from the NN-XT. Turn the knob to the right to increase the volume.

Overview of the Remote Editor panel

It is in the Remote Editor Panel that the main NN-XT action is going on, especially if you're creating your own patches. The remote editor is dominated by the key map display, and this is also the part on which we will concentrate to begin with.

The Key Map display

The key map display consists of a number of separate areas that let you do different things. To help you navigate the key map display, these areas are described below.



The Info area

This displays the following information about the currently selected sample: Sample rate, mono/stereo information, bit resolution and file size.

The Sample area

This area displays the names of the samples in each zone. It also allows you to change the order of the zones by clicking and dragging them up and down.

The Group area

This area does not show any information. However, by clicking in it, you can instantly select all the zones that belong to a certain group. See "Working with Grouping" for information on how to create groups.

The Keyboard area

Aside from the fact that is a guideline for setting up key ranges, it is also used for setting the root keys of, and auditioning loaded samples. See "About the Root Key" and "Auditioning samples" respectively for more information.

The Tab Bar area

This area gives you a visual indication of the key range of a selected zone. By clicking and dragging the "handles" at the key range boundaries, you can resize the key ranges, and by clicking in between the handles, you can move the key ranges without changing their length.

The Key Range area

This area in the middle of the key map display is where you keep track of all the zones and the relationship between them. You can also move and resize the zones just like in the Tab Bar area, as described above.

The Scrollbars

There are both horizontal and vertical scrollbars that work just like regular scrollbars. Whenever there is more information in the key map display than what fits on a "single screen", you can use the scrollbars to reveal it. Either click on the arrows or click and drag the scrollbar handles.

Sample parameters



This area shows the current values of basic parameters you can set for zones, such as root key, play mode, output etc. The parameters are changed by using the knobs directly below the key map display.

Group parameters



These parameters are adjusted on a per group basis (see "Group parameters" for more information on groups). Most of them relate to performance or playing style.

Synth Parameters



The bulk of the parameters on the remote editor are used for adjusting the sound of the samples by applying filtering, envelope shaping, modulation (like vibrato and tremolo) and so on. We call these the synth parameters, since they are to a large extent identical to those on a regular synthesizer.

About Samples and Zones

For a clear understanding of the terminology used when describing the various operations that can be performed in the key map display, it is important to clarify the distinction between a *sample* and a *zone*:

- **A Sample is a piece of audio that can be loaded into the NN-XT and played back.**
- **A Zone could be viewed as a "container" for a loaded sample.**

All loaded samples are placed in "Zones" in the key map display. You can then organize the zones as you please, and make various settings such as key- and velocity ranges separately for each zone.

In other words, the settings you make are actually performed on the zones, but affect the samples in them. Hence, when we talk about making settings for a zone, it is synonymous with making settings for a sample - the sample that the zone contains.

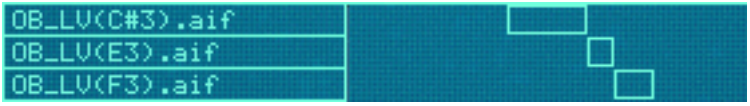
- **Two or more zones can play the same sample, but with different parameter settings, making them sound completely different.**
- **A zone can be empty, playing no sample at all.**

Selections and Edit Focus

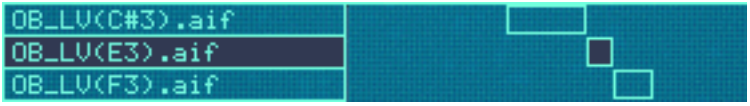
Almost all operations in the remote editor are performed on one or more selected zones or on the zone with edit focus. Several zones can be selected at once, but only one zone at a time can have edit focus.

This is important since:

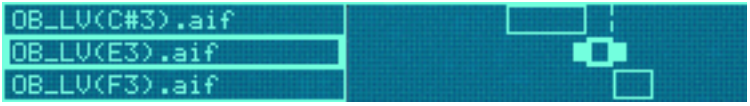
- **Editing operations that can be performed on several zones (like deleting), always apply to the selected zones.**
- **Editing operations that can be performed on one zone only, always apply to the zone with edit focus.**
- **The front panel always shows the settings for the zone with edit focus.**



Here no zone is selected.



Here the middle zone is selected but does not have edit focus.



Here the middle zone has edit focus but is not selected. Notice the thicker border and the additional handles in the key range area.



Here the middle zone is selected and has edit focus.



Here, all three zones are selected, but the middle one has edit focus.

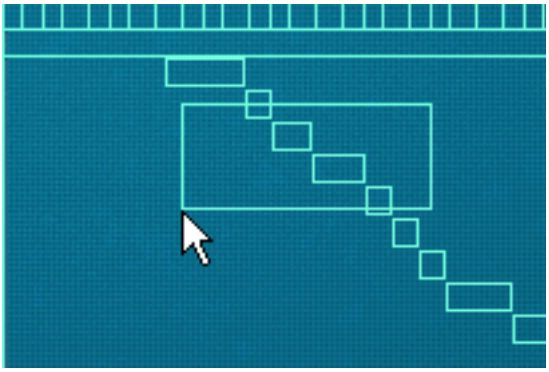
Selecting Zones

- **To select a zone, click on it.**
 Clicking on a zone will also automatically give it edit focus.

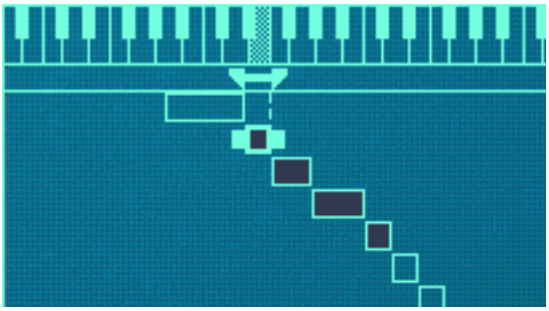
You can also select multiple zones in several ways:

- **By holding down [Shift], or [Ctrl](Win) or [Cmd](Mac) and clicking on the zones you wish to select.**
 This way you can select several non-contiguous zones. You can also deselect a selected zone by clicking on it again.
- **By clicking and dragging a selection box in the key map area.**

Making a selection box like this....



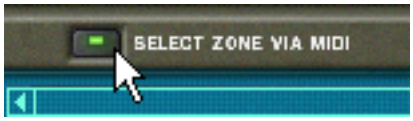
...will select these zones:



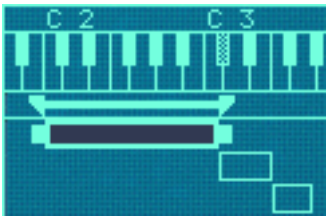
Note that the zones don't have to be completely encompassed by the selection box. The selection box only have to intersect parts of the zones to include them in the selection.

Selecting zones via MIDI

You can also select zones via your MIDI keyboard. By clicking the button marked “Select zones via MIDI” above the key map display so that it lights up, you enable selection via MIDI.



This way, you can select a zone and give it edit focus by pressing a key that lies within the zone’s key range (see later in this chapter for information about setting up key ranges).



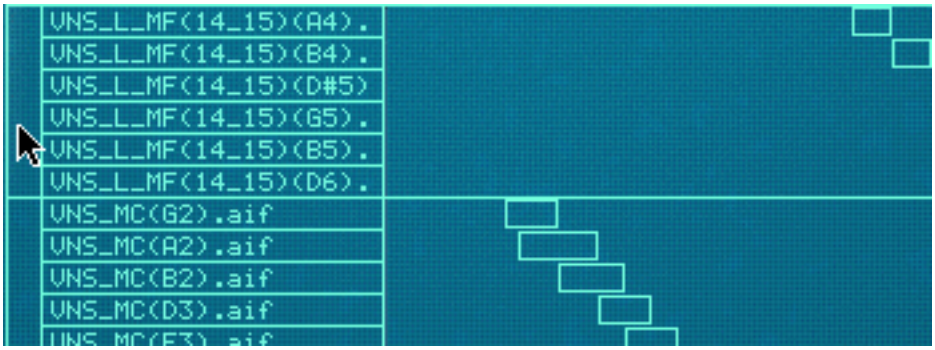
In this case, this zone can be selected by pressing any key between C2 - C3 on your MIDI keyboard.

Note also, that selection via MIDI is velocity sensitive. Zones may have specific velocity ranges. This means that they won’t be played unless the key that triggers the zone is played with a certain velocity. The same rules apply when selecting via MIDI, only zones that meet the velocity criteria will be selected. Read more about setting up velocity ranges on “Setting velocity range for a Zone”.

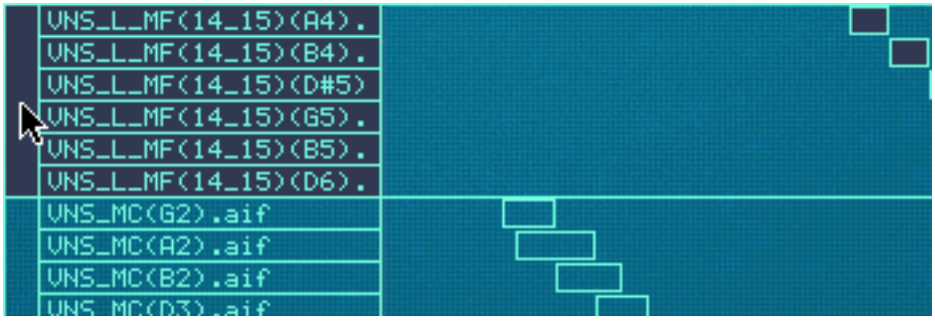
Selecting All Zones in a Group

The concept of zone groups is fully introduced on “Working with Grouping”. For now we will only describe how to select all samples that belong to the same group:

Clicking in the Group column...



...selects all zones in the group



Moving Edit Focus

Moving Edit Focus

A zone can be given edit focus independently of selection:

- **When you click on an unselected zone, it both gets selected and gets edit focus.**
- **When you select several zones using [Shift], or [Ctrl](Win) or [Cmd](Mac), the one you select last always gets edit focus.**
- **To set edit focus to a zone when several zones are already selected, click on it without holding down any modifier keys.**
 This way, you can move edit focus between the selected zones freely without deselecting any of them.

Adjusting parameters

Adjusting Synth parameters

The synth parameters are the ones that occupy the bulk of the remote editor panel (see "Synth Parameters"). Changes you make to synth parameters always apply to all selected zones.

- **The panel always shows the settings for the zone with edit focus.**
 More about this below.
- **To make adjustments to one zone, select it (which also gives it edit focus) and adjust the parameter on the front panel.**
- **To set *several* zones to the same value, select them and adjust the parameter.**
 All zones will be set to the same value for the parameter you adjusted.

Adjusting Group parameters

Group parameters apply to a group. That is, they are settings that are shared by all zones in a group.

- **To make adjustments to one group, select one or more zones that belong to the group, and adjust the parameter on the front panel.**
- **To set several groups to the same value, select at least one zone in each group you want to adjust, and adjust the parameter.**
 All groups will be set to the same value. More about this below.

About "Conflicting" parameters

Often you will find yourself in a situation where you select multiple zones and parameter settings differ between them. This is quite normal. For example, you will often find yourself making adjustments to for example level and filtering to balance the sound between several samples across the keyboard. However, if you have multiple selections this can sometimes lead to confusion: Enter the NN-XT's "conflicting parameters" indication:

Whenever two or more *selected* zones have conflicting parameter settings, NN-XT will notify you about this by showing a small "M" (for multiple) symbol, next to the parameter.



In this example, Level and Spread have conflicting settings.

- **The controls on the panel always show the setting for the zone with edit focus.**
- **By clicking your way through the zones within the selection, you can see the settings for each zone.**
- **If you adjust a parameter, all selected zones will be set to the same value for this parameter.**

You can put this functionality to good use when checking how a patch has been created and when checking that your own settings are consistent through the various zones.

Sample parameters

The Sample parameters allow you to specify various properties for one or several selected zones, such as tuning, key and velocity ranges.



- **To set several zones to the same value, select them and adjust the parameter.**
 All zones will be set to the same value for the parameter you adjusted.

Copying parameters between zones

You can easily copy parameter settings from one zone to any number of other zones. Proceed as follows:

1. **Select all the zones you want to involve in the operation.**
 By this we mean the zone with the settings you wish to copy, and the zone(s) to which you want to copy the settings.
2. **Make sure the zone that contains the settings you want to copy has edit focus.**
3. **Pull down the Edit menu or the NN-XT context menu and select "Copy Parameters to Selected Zones".**
 All the selected zones will now get the exact same parameter settings.

! **Observe that this only applies to the synth parameters (see "Synth parameters"). Sample parameters (root key, velocity range etc.) can not be copied.**

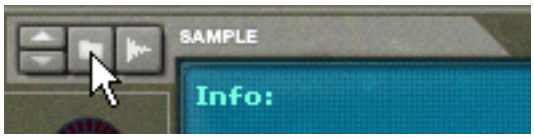
Managing Zones and Samples

Creating a Key Map

When you add an NN-XT sampler to the rack and select “Reset Device” from the context menu or Edit menu, its key map display becomes empty. That is, it contains no samples.

To create a new key map, proceed as follows:

1. **Either click the Browse Samples button, select Browse Samples from the Edit menu or select Browse Samples from the NN-XT’s context menu.**
 The NN-XT device gets browse focus.



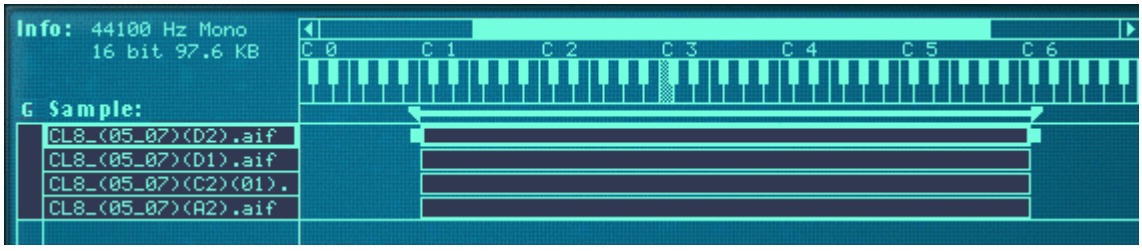
The Browse Samples button.

2. **Select the sample or samples that you want to load in the browser and click the “Load” button in the Browser.**
 The selected sample(s) are loaded into the NN-XT.

→ **Alternatively, drag one or several sample files from the Browser and drop it on the NN-XT device in the rack.**
The panel is dimmed in blue and a Sample Replace symbol appears in the center.

When new samples are loaded into the NN-XT they have the following properties:

- **Each sample is placed in its own zone.**
- **Each zone spans a key range of five octaves on the keyboard - C1 to C6.**
- **All the newly added sample(s)/zones are automatically selected.**
- **The first added zone gets edit focus.**



A key map with four newly added samples.

Setting Root Notes and Key Ranges

The next step after loading the samples is most likely to adjust the key range, root note and tuning of the samples, so that they play sensibly across the key range. There are many ways of doing this, described in “Working with Key Ranges” and onwards. However, we will here briefly describe a procedure for quickly creating a complete key map out of a set of loaded samples.

This example assumes that the samples you load is a set of multisamples for a pitched instrument (like guitar, piano, flute etc.).

1. **Load the samples.**
2. **Use “Select All” on the Edit menu to select all the loaded samples.**
3. **Use “Set Root Notes from Pitch Detection” to automatically set up the root notes (pitches) for the samples.**

4. **Select "Automap Zones" from the Edit menu.**

All selected zones are automatically arranged into a basic key map. You can now proceed with adjusting the synth parameters on the front panel to shape the sound!

About file formats and REX slices

The audio file format support differs depending on which computer OS you are using.

The NN-XT can read audio files in the following formats:

- **In Windows:**
 .wav, .aif, .mp3, .aac, .m4a and .wma.
- **In macOS:**
 .wav, .aiff, .3g2, .3gp, .mp1, .mp2, .mp3, .mpeg, .mpa, .snd, .au, .sd2, .ac3, .aac, .adts, .amr, .caf, .m4a .m4r and .mp4.
- **SoundFont samples**
 This is a standardized data format containing wavetable synthesized audio and information on how it should be played back in wavetable synthesizers - typically on audio cards. SoundFont banks are hierarchically organized into different categories: User Samples, Instruments, Presets etc. The NN-XT lets you load single samples from within a Soundfont bank.
- **REX file slices**
 A slice is a snippet of sound in a REX File (see "Loading complete REX files as Patches"). To import a REX slice, browse to a REX file and open it as if it was a folder. The browser will then display the slices as files inside that "folder". In the rest of this manual, when we refer to importing samples, all that is said applies to REX slices as well.
- **Any sample rate and practically any bit depth.**

Adding more samples to the Key Map

You can add additional samples to an existing key map way described above.

1. **Make sure that no already loaded sample has edit focus.**
 If you don't, there's a risk that the selected sample will be replaced, see below. To remove the edit focus, click in an unoccupied area in the Sample column or the key map area.
2. **Open the Sample Browser.**
3. **Select the sample(s) you want to load in the browser and click the "Load" button in the Browser.**

→ **Alternatively, drag one or several sample files from the Browser and drop it on the NN-XT device in the rack.**
The panel is dimmed in blue and a Sample Replace symbol appears in the center.

The new sample(s) are added to the key map.

Replacing a sample

To replace the sample in a zone, proceed as follows:

1. **Make sure the zone has edit focus and do one of the following:**

→ **Click the Browse Samples button.**

→ **Select Browse Samples from the Edit menu or the NN-XT context menu.**

→ **Double click in the zone.**
Any of these methods will set browse focus and open the standard file browser in which you can select new samples for the zone.

2. **Select one and only one sample in the Sample Browser.**
 If you select more than one sample in the browser the samples you load will not replace the one with edit focus. They will instead be added below it.
3. **Click the Load button in the Browser.**

→ **Alternatively, drag a sample file from the Browser and drop it on the NN-XT device in the rack.**
The panel is dimmed in blue and a Sample Replace symbol appears in the center.

Quick browsing through samples

If you want to quickly browse through a number of samples, for example to see which one of them would fit best in a certain context, proceed as follows:

1. **Set up the zone as desired and make sure it has edit focus:**
2. **Use the arrow buttons in the Browse Samples section to select the next/previous sample in the directory.**

Removing samples

→ **To remove a sample from a zone, select it by clicking on it and then select "Remove Samples" from the Edit menu or the NN-XT context menu.**
This will remove the sample from the zone, leaving it empty. Note that you can remove the samples from several selected zones at the same time.

Auditioning samples

You can audition the loaded samples in two ways:

→ **By pressing [Alt](Win) or [Option](Mac) and clicking a sample in the sample column.**
The mouse pointer will take on the shape of a speaker symbol when you move it over the sample column.

Clicking a sample will play it back at its root pitch (see "About the Root Key"). Furthermore, the sample will play back in its unprocessed state. That is, without any synth-parameters applied (see "Synth parameters").

→ **By pressing [Alt](Win) or [Option](Mac) and clicking a sample in the keyboard column.**
The difference here is that you will hear the sample at the pitch corresponding to the key you clicked and with any and all processing applied. The click mimics a key played with velocity 100. Also note that this may trigger several samples, depending on whether they are mapped across the same or overlapping key ranges, and the velocity range settings (see "Setting up Key Ranges" and "Setting velocity range for a Zone" respectively).

Adding empty Zones

You can add empty zones to a key map. Empty zones are treated just like zones containing samples, in that they are automatically selected, gets edit focus and are assigned a five octave key range when they are first created. However, you can only add one zone at a time. It is also possible to resize, move and edit empty zones in the same way as zones containing samples.

→ **To add an empty zone, pull down the Edit menu or the NN-XT context menu and select "Add Zone".**
An empty zone is added below any existing zones in the key map. An empty zone is indicated with the text "**No Sample**".

After you have added an empty zone, you can assign a sample to it, just as when Replacing a Sample, or when Quick Browsing, as described above.

Duplicating Zones

You can duplicate any number of already existing zones (containing samples or empty).

1. **Select the zone(s) you want to copy.**
2. **Pull down the edit menu or the NN-XT context menu and select "Duplicate Zones".**
 The selected zones will now be copied and automatically inserted below the last one in the key map display.

The duplicated zones will contain references to the same samples as the original zones. They will also have the exact same key ranges and parameter settings.

Using Copy and Paste

The Copy Zones function on the Edit menu allows you to copy all selected zones to the clipboard. Selecting Paste Zones from the Edit menu will paste the zones into the selected NN-XT device, below the existing zones.

This is a handy way to transfer zones (complete with all settings) from one NN-XT device to another.

Removing Zones

To remove one or several zones, select them and do one of the following:

- **Press [Delete] or [Backspace] on the computer keyboard.**
- **Select "Delete Zones" from the Edit menu or the NN-XT context menu.**

When removing zones, you will remove any samples in them as well.

Rearranging Zones in the List

- **To move a zone to another position in the list, click on it in the samples column and drag up or down.**
 An outline shows you where the zone will appear when you release the mouse button.

Working with Grouping

About Groups

Grouping has two purposes:

- **To allow you to quickly select a number of zones that "belong together."**
 For example if you have created a layered sound consisting of piano and strings, you could put all string samples in one group and all piano samples in one group. Then you can quickly select all piano samples and make an adjustment to them by trimming a parameter.
- **To group zones that need to share group settings together.**
 For example, you may want to set a group to legato and monophonic mode and add some portamento so that you can play a part where you slide between notes.

Note that there is always at least one group, since the zones you create are always grouped together by default.

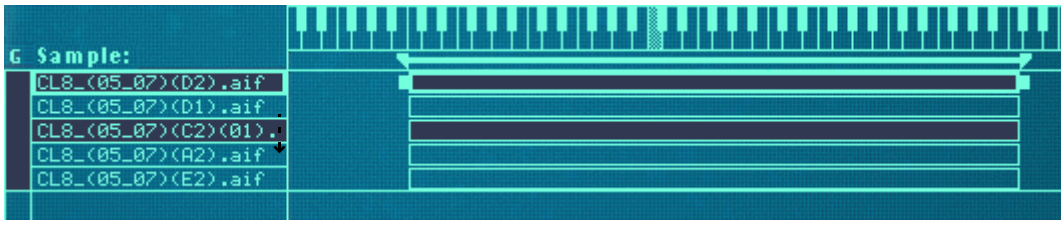
Creating a Group

1. **Select the zones you want to group together.**
 The zones don't have to be contiguous in order to be grouped. Regardless of their original positions in the samples column, they will all be put together in succession.

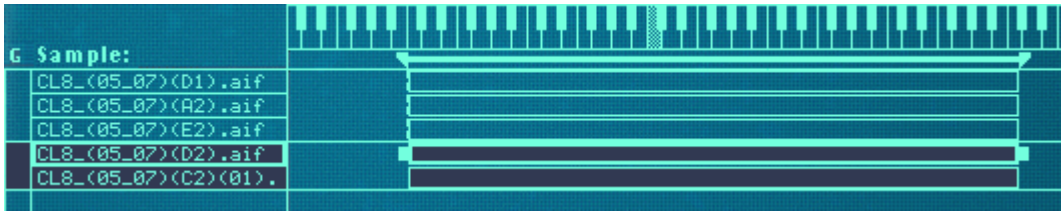


2. **Select "Group Selected Zones" from the Edit menu or the NN-XT context menu.**
 The zones are grouped.

 Selecting these zones and grouping them...

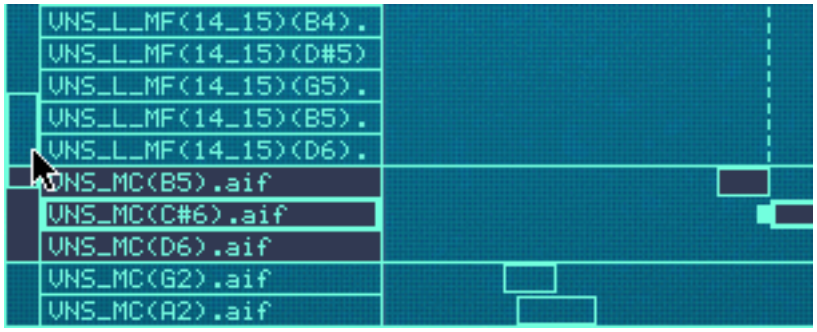


 ...will create these two groups instead of the original one large group.



Moving a Group to another position in the List

→ **Click on the group in the Groups column and drag up or down with the mouse button pressed.**
An outline of the group you move is superimposed upon the display to help you navigate to the desired position.



Dragging a group to a new position.

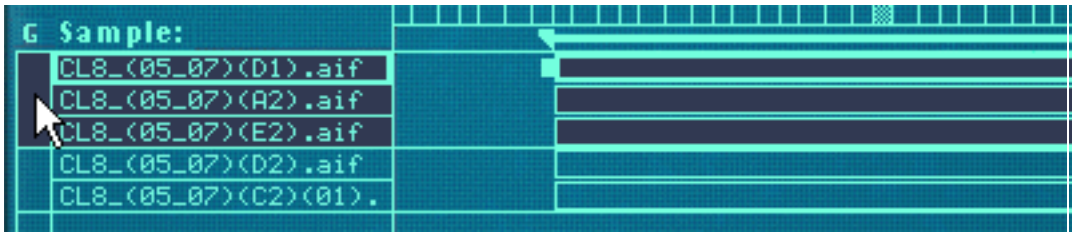
3. **Release the mouse button at the desired position.**
 The group and all its zones appear at the new position.

Moving a Zone from one Group to another

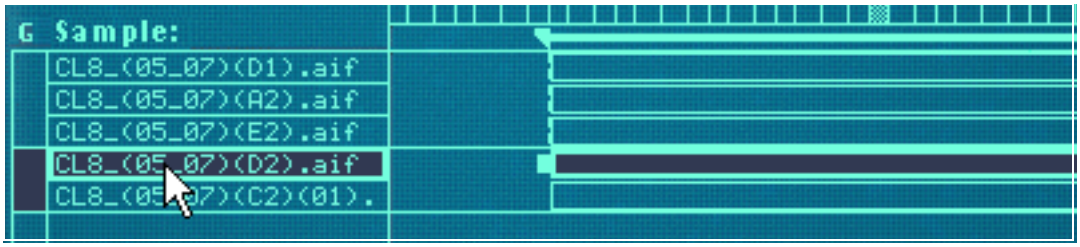
This is done just as when rearranging samples in the list, as described on the previous page. The only difference is that you drag the zone from one group to another.

Selecting a Group and/or Zones in a Group

→ **Clicking on a group in the groups column selects the group and all the zones in the group.**



→ **Clicking on a zone in the samples column selects the group (and that zone).**



The Group Parameters

There are a few parameters on the front panel that apply specifically to groups. see "Group parameters" for details.

Working with Key Ranges

About Key Ranges

Each zone can have its own separate key range, the lowest and the highest key that will trigger the sample.

A good example of use for this is when sampling a certain instrument. Sampling of a piano for example is usually performed by making several recordings of different notes at close intervals, and then mapping these samples to separate, contiguous, fairly narrow key ranges. This concept is called multi-sampling.

The reason for this is that if one single sample is played across the entire keyboard, it will most likely sound very unnatural when played too far from its original pitch, since the amount you can transpose a sound without negatively affecting its timbre is very limited.

Setting up Key Ranges

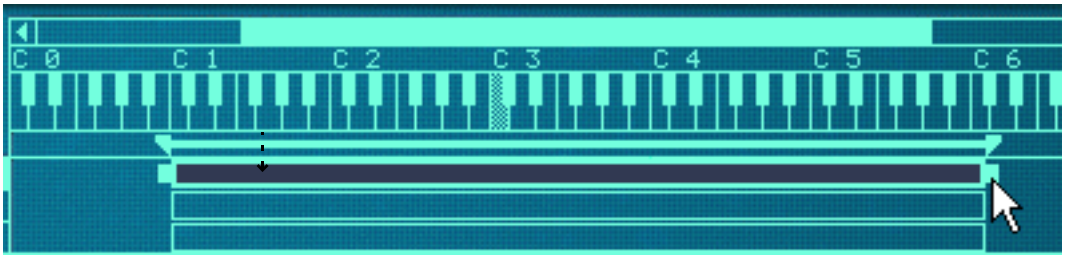
You can adjust the key range of zones in a number of ways:

By Dragging the Zone Boundary Handles

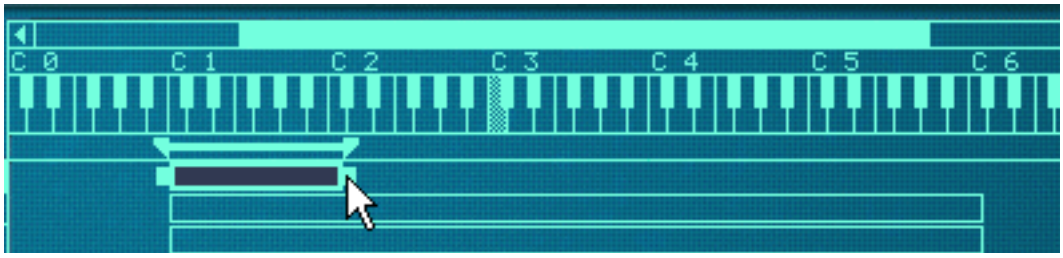
1. **Select the zone in the Key Range area.**
2. **Point and click on one of the handles that appear at each end.**

3. **Drag the handle left/right.**
 Dotted lines extend from the edges of the zones up to the keyboard area. These lines give you a visual indication of which keys the key range will encompass. There is also an alphanumerical indication at the bottom left of the display.

Clicking and dragging the high key boundary handle of a zone with the default key range of C1 - C6...



...to change the key range to C1 - C2.



4. **Repeat the procedure with as many zones as you wish, to create a complete key map.**

By using the Lo Key and Hi Key controls

Directly below the key map area you will find the sample parameters. These are used for changing various parameters that affect how the zones are played back. They can affect single or multiple selected zones. In the middle of the sample parameters area are two knobs called “Lo Key” and Hi Key”.



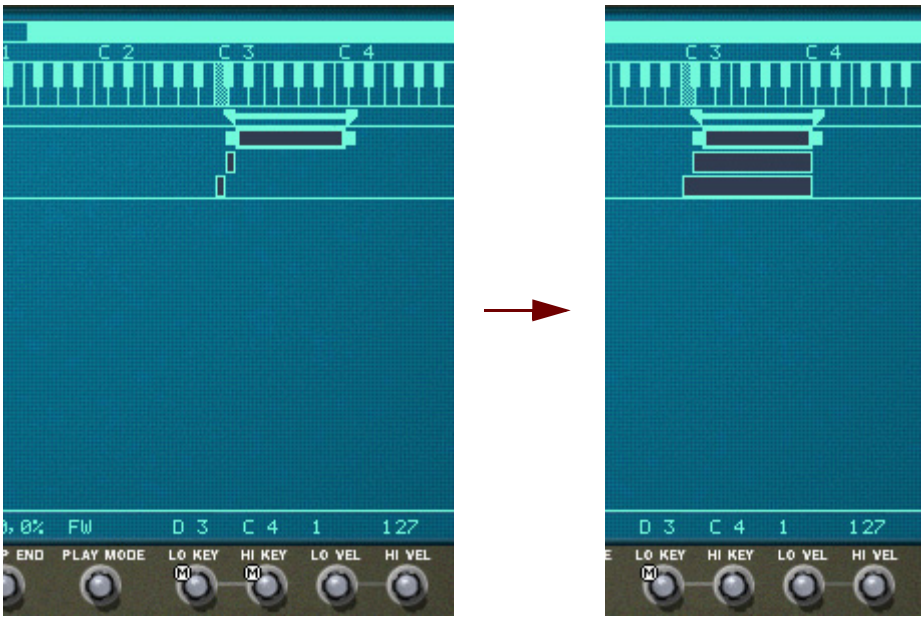
These can be used for setting the low key and the high key of a zone’s key range.

1. **Make sure the zone which you want to set the key range for is selected.**
2. **Use the Lo Key/Hi Key knobs to change the key range.**
 Check the display right above the knobs for an indication of the key. You can also keep an eye on the lines extending from the zone edges to the keyboard area.

Setting key ranges for multiple zones

You can set key ranges for multiple selected zones simultaneously. This can only be done by using the Lo and Hi Key controls. It works as follows:

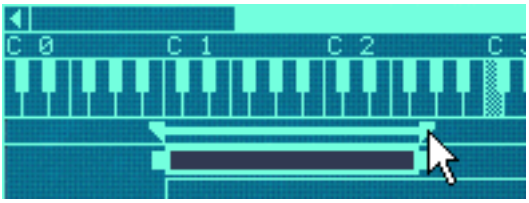
- **If you have three selected zones that each have different high keys and then turn the Hi Key knob, they will all automatically get the same High Key value as the zone with edit focus.**
 In other words, if the selected zone with edit focus has the high key set to C4, and you change this to D4 by turning the Hi Key knob, all other selected zones will also be extended to D4 as the High Key.



- **If any selected zone's low key setting is higher than the edit focused zone's high key before turning the Hi Key knob, the zone range will be scaled down to one semitone, starting from the low key setting.**
 The high key can naturally never be set to a value lower than one semitone above its low key setting - the zone would otherwise disappear!
- **The inverse is also true - i.e. turning the Lo Key knob for several selected zones will apply the edit focused low key setting to all selected zones.**
 A low key can never be set higher than one semitone below the high key in a zone, so if the edit focused zone has a low key above the high key of another zone, the other zone will be scaled to the minimum semitone range.

By Dragging the Zone Boundary Handles on the Tab Bar

As previously described, the area directly below the keyboard area is called the tab bar. This shows the key range for the currently selected zone, and also contains boundary handles.



Dragging a boundary handle on the tab bar.

These handles can be used much to the same effect as when dragging the boundary handles in the key map display. However, the handles on the tab bar can change the key range of multiple zones at the same time.

The following applies:

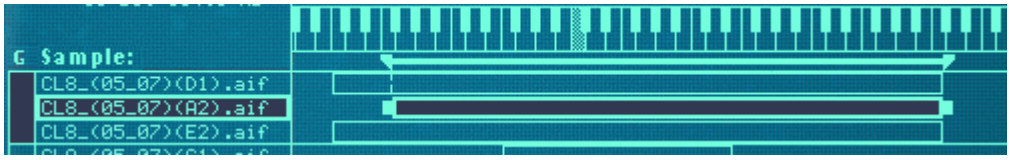
- **The tab bar shows the key range for the zone with edit focus.**

- **Dragging the boundary handles for that zone will also simultaneously change the key range for a number of surrounding zones if:**

→ **The high key or low key (depending on which handle you drag) of the other zones are the same as the zone with edit focus.**

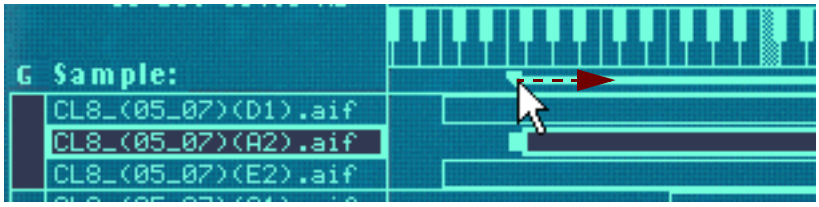
→ **The other zones are adjacent to the zone with edit focus.**

! **Note that it doesn't matter whether the other zones are selected or not. They will be affected anyway.**



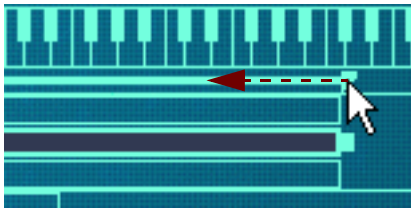
In the example in the picture above, the zone in the middle has edit focus. Its left handle (the low key) is placed differently from any of the other zones, but all of the zones have the same high key setting. This means that...

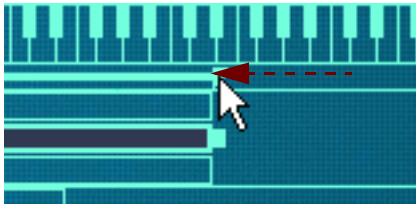
- **Dragging the left handle will only move the low key position of the zone with edit focus (the pictures show before and after dragging).**





- **Dragging the right handle will move the high key position for all of the zones at the same time, since they all have the same high key position (again, the picture shows before and after dragging).**

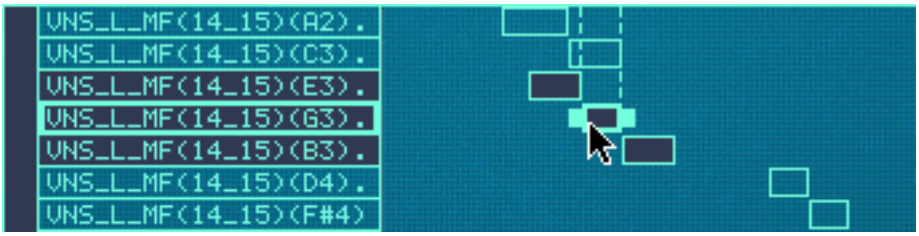




Moving Zones by Dragging the Zone Boxes

You can also move entire zones horizontally, thereby changing their key ranges.

1. **Select all the zones you want to move.**
 You can move several zones simultaneously.
2. **Point on any of the selected zones, and press the mouse button.**
3. **Drag left/right and release the mouse button.**

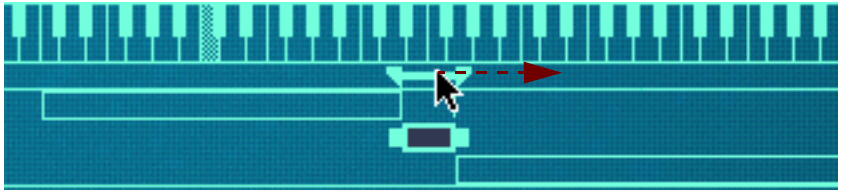


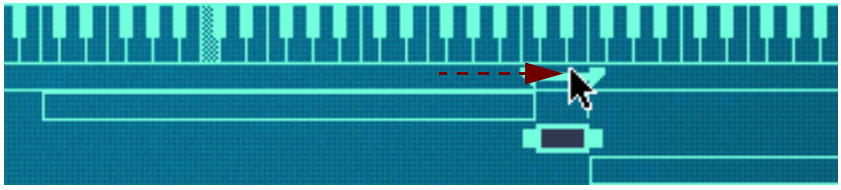
Dragging multiple zones.

Moving Zones by Dragging in the Tab Bar

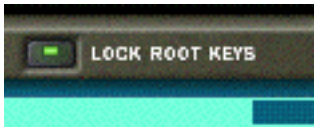
You can also move a zone by dragging anywhere between the zone boundary handles on the tab bar. When you do, the surrounding zones will be affected just as when dragging the boundary handles in the tab bar (see above).

This can be used to "slide" a zone in relation to its surrounding zones, as the picture example below shows (before and after dragging).





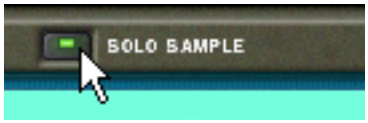
About the Lock Root Keys function



Normally, when you move zones (as described above), the root note of the zone(s) you move will change accordingly. In other words, the zone(s) will be transposed. If this is not desired, you can activate the Lock Root Keys function prior to moving the zone(s) by clicking on the button above the key map display.

Moving zones without changing their root notes can be used for some interesting effects, since it will completely change the timbre of the sample(s) as they are played back.

About the Solo Sample function



The Solo Sample function lets you play a selected sample over the entire keyboard and disregarding any velocity range assigned to the sample. All other loaded samples are temporarily muted.

This is useful if you for example want to check how far up and down from its root key a sample can be played on the keyboard before starting to sound "unnatural". The solo sample function can therefore be useful as a guide for setting up key ranges, as described in "Setting up Key Ranges".

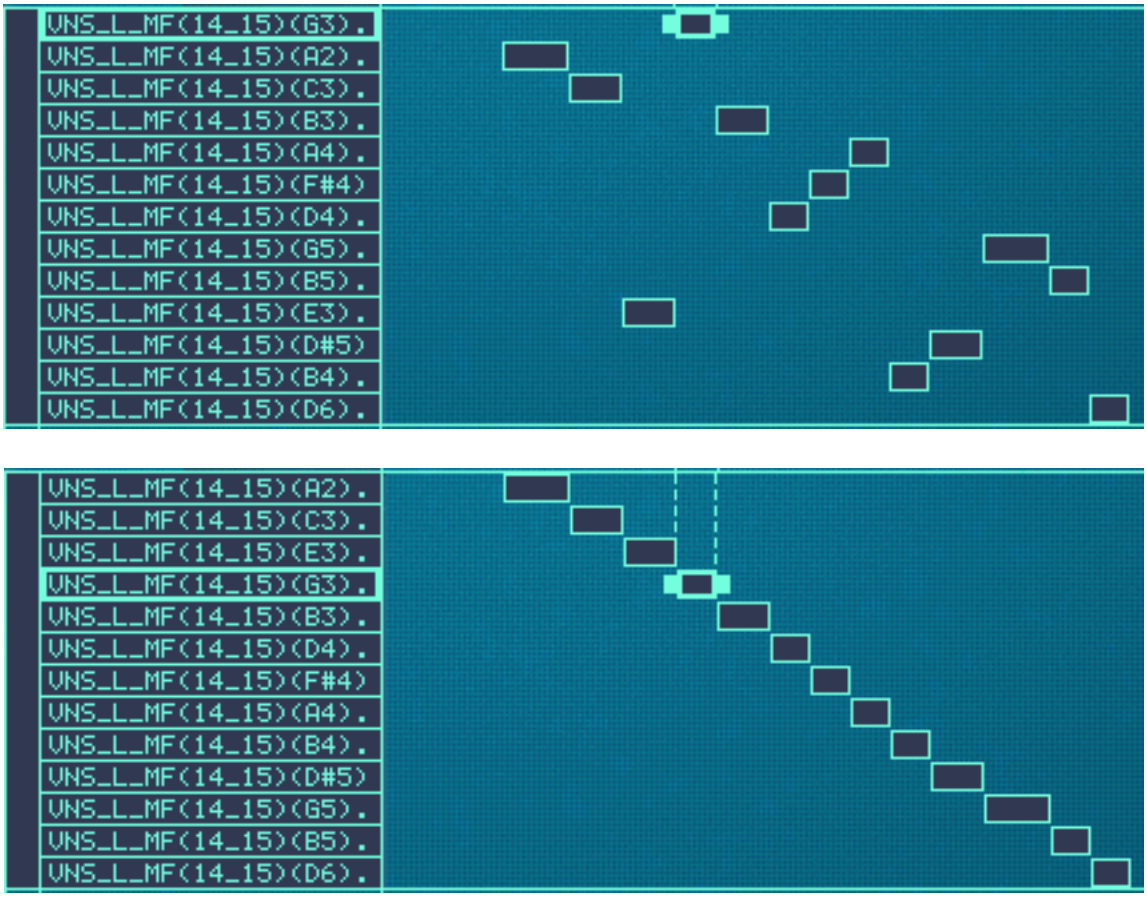
1. **Select one and only one zone, or - if you have a selection of multiple zones - make sure the one you want to hear has edit focus.**
2. **Activate Solo Sample by clicking on the button so that it lights up.**
3. **Play the MIDI keyboard**

Sorting Zones by Note

The Edit menu and the NN-XT context menu contains an item called “Sort Zones by Note”. This option lets you automatically sort the selected zones in descending order according to their key ranges.

When you invoke this option, the selected zones will be sorted from top to bottom in the display starting with the one with the lowest range.

Note however, that the sorting is done strictly on a group basis. That is, only zones that belong to the same group can be sorted in relation to each other.



Before sorting and after.

If two zones have the same key range, they are sorted by velocity range.

Setting Root Notes and Tuning

About the Root Key

All instrument sounds have an inherent pitch. When playing a sample of such a sound on the keyboard, the keys you play must correspond to that pitch. For example, you may have recorded a piano playing the key "C3". When you map this onto the NN-XT key map, you must set things up so that the sampler plays back the sample at original pitch when you press the key C3.

This is done by adjusting the root note.

- **Many samples files from different sources already have a set root key in the file. If they do, the root key will be correctly set automatically when you load the sample into a zone.**
- **However if the sample doesn't have a root note stored in the file, (if you for example have recorded it yourself) you will need to adjust it**

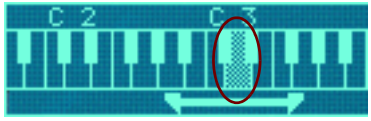
Setting the Root Note manually

To set the root key for a zone, proceed as follows:

- **Make sure the zone has edit focus (for example by clicking on it), and do one of the following:**
- **Use the knob marked "Root" in the sample parameter area below the display.**
 Turning it to the right will raise the pitch of the root key. The selected key is displayed alphanumerically directly above the knob, and you can also look at the keyboard area for a visual indication (see below).



- **Press [Ctrl](Win) or [Cmd](Mac) and click on the desired root key in the keyboard area.**
 The set root key is shaded so you can easily distinguish it.



Tuning samples manually

In addition to setting the root note, you may need to fine tune your samples, in order for them to match other instruments and/or each other:

- **Make sure the zone has edit focus (for example by clicking on it).**
- **Use the knob marked "Tune" in the sample parameter area.**
 This allows you to tune each sample in a key map by +/– half a semitone (-50 – 0 – 50).



Setting the Root Note and Tuning using pitch detection

The NN-XT features a pitch detection function to help you set the root keys. This is useful if you for example load a sample that you haven't recorded yourself, and you don't have any information about its original pitch.

Proceed as follows:

1. **Select all the zones you want to be subject to pitch detection.**
2. **Pull down the Edit menu or the NN-XT context menu and select "Set Root Notes from Pitch Detection".**
 The samples in all the selected zones will now be analyzed, and the detected root keys will automatically be set for you.

! **Note that for this to work properly, the samples must have some form of perceivable pitch. If it is sampled speech, or a snare drum for example, it probably doesn't have any discernible pitch.**

About changing the pitch of samples

The procedures above should be used to make sure the samples are consistently pitched across the keyboard, and that they all match an absolute reference (for example A 440 tuning).

If you need to tune the samples to match other material, or to get a certain effect (for example detuning two sounds against each other for a chorus effect) you should use the Pitch section among the synth parameters, not the sample tuning parameters.

Using Automap

The automap function can be used as a quick way of creating a key map, or as a good starting point for further adjustments of a key map.

Automap works under the assumption that you intend to create a key map for a complete instrument, for example a number of samples of a piano, all at different pitches.

1. **Load the samples you want to Automap.**
 Now you have three options:

→ **Trust that the root note information in the files is already correct.**

→ **Manually adjust the root notes (and tuning) for all the samples.**

→ **Use "Set Root Notes from Pitch Detection" to automatically set up the root notes.**

2. **Select all zones you want to automap.**
3. **Select Automap Zones from the Edit menu or the NN-XT context menu.**

All the selected zones will now be arranged automatically in the following way:

→ **The zones will be sorted in the display (from top to bottom - lowest key first) according to the root keys.**

→ **The zones will be assigned key ranges according to the root keys.**
The key ranges are set up so that the split between two zones is exactly in the middle between the zones' root notes. If two zones have the same root key they will be assigned the same key range.

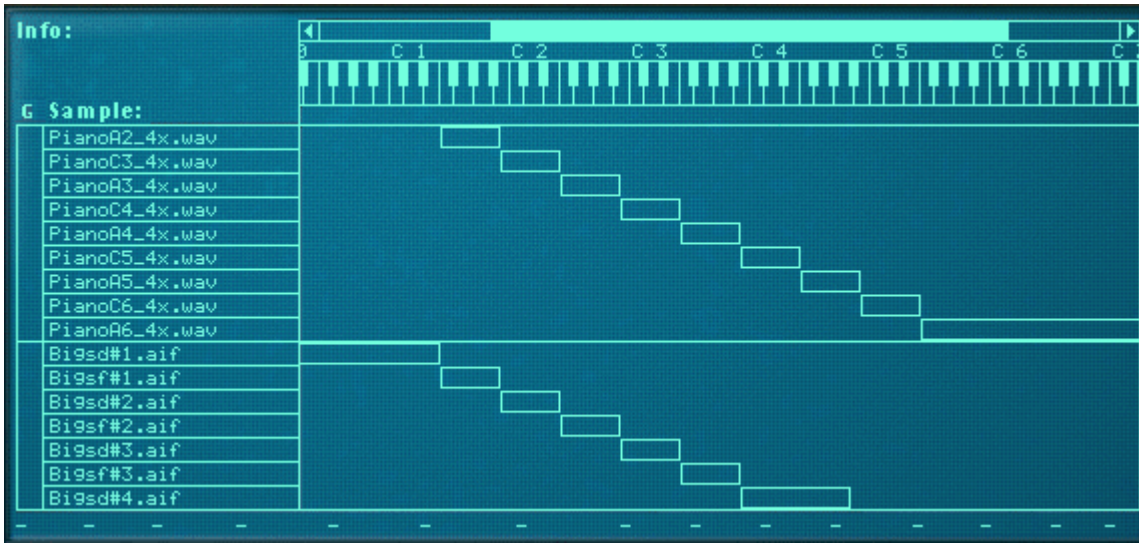
Automapping zones chromatically

→ **This Edit menu item will give each zone a key range of one semitone (i.e. one key), starting from C2 and upwards.**
The function does not take root key into account. It simply places each selected sample on successive keys according the position in the sample list (from the top down).

Layered, crossfaded and velocity switched sounds

Creating layered sounds

You can set things up so that two or more zones have overlapping key ranges - either completely or partially. This way you can create layered sounds, i.e. different samples that are played simultaneously when you press a key on your keyboard.



In the picture above, you can see a set of piano samples at the top, mapped across the key range.

Below these are a set of string samples that also span the entire key range.

Whenever you play a key within this keyboard range, the sound produced will be a combination of the piano and the string sample.

In addition, in the example above, the user has arranged the piano samples into one group and the string samples in another. This is convenient since it allows for quick selection of the entire piano map, for example for balancing its level against the strings.

About velocity ranges

When zones are set up so that their key ranges overlap – completely or partially – you can use velocity switching and crossfading to determine which zones should be played back depending on how hard or soft you play on your MIDI keyboard.

This is done by setting up velocity ranges, with or without crossfading.

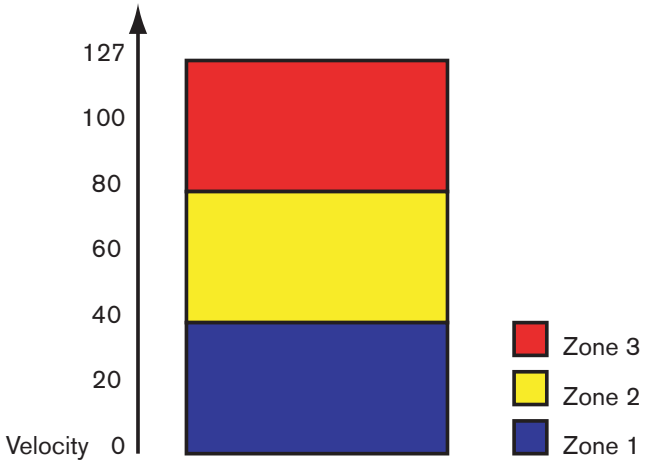
Each time you press a key on your MIDI keyboard, a velocity value between 1-127 is sent to Reason. If you press the key softly, a low velocity value is sent and if you press it hard, a high velocity value is sent.

This velocity value determines which samples will be played and which will not.

Let's say for example that you've mapped three different zones across the same key range:

- **Zone 1 has a velocity range from 1-40.**
 This means that the sample in it will be triggered by velocity values between 1-40.
- **Zone 2 has a velocity range of 41-80.**
 The sample in this zone will be played back by velocity values between 41-80.

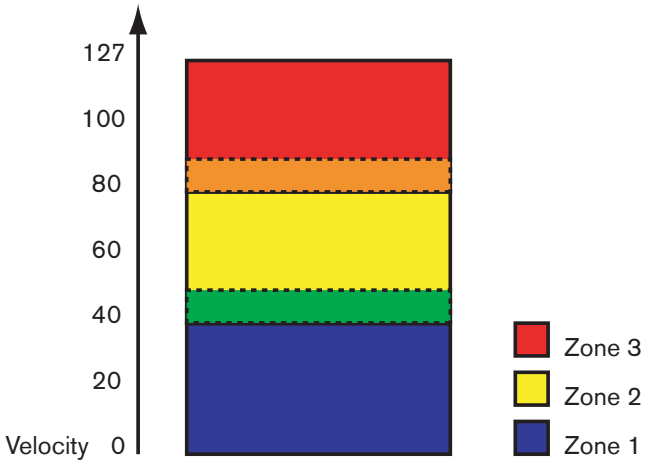
- **Zone 3 has a velocity range of 81-127.**
 The sample in this zone will be triggered by all velocity values above 80.



Overlapping velocity ranges

Let's change the values above slightly:

- **Zone 1 has a velocity range from 1-60.**
- **Zone 2 has a velocity range of 41-100.**
- **Zone 3 has a velocity range of 81-127.**

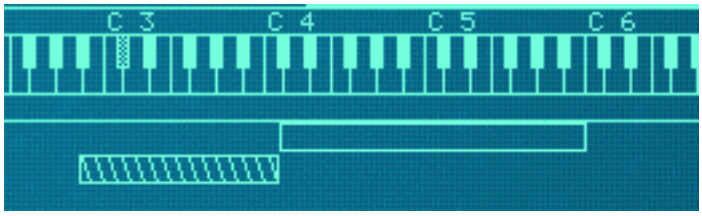


Now, velocity values between 41 and 60 will trigger samples from both Zone 1 and Zone 2. Likewise, velocity values between 81 and 100 will trigger sounds from Zone 2 and Zone 3.

About full and partial velocity ranges

You can see which zones have modified velocity ranges in the key map display:

- **Zones with a full velocity range (0 - 127) are only shown with an outline.**
- **Zones with any other velocity range are shown as striped.**



The top zone has a full velocity range (1-127), and the lower zone has a partial velocity range (any other range), which is indicated by stripes

Sorting Zones by velocity values

The Edit menu and the NN-XT context menu contain an item called "Sort Zones by Velocity". This option lets you automatically sort the selected zones in the display in descending order according to their set low or high velocity values.

When you invoke this option, the selected zones will be sorted from top to bottom starting with the one with the highest "Lo Vel "value.

Note however, that the sorting is done strictly on a *group* basis. That is, only zones that belong to the same group can be sorted in relation to each other.

If two zones have the same velocity range, they are sorted by key range.

Setting velocity range for a Zone

To set up a velocity range for a zone, proceed as follows:

1. **Select one or more zones that you want to adjust.**
2. **Use the knobs marked "Lo Vel" and "High Vel" in the sample parameter area to set the desired low- and high velocity values.**



Adjusting the "Lo Vel" value for a zone.

"Lo vel" is the lowest velocity value that should trigger the sample in the zone - i.e. if a key is pressed so softly that the velocity is lower than this value, the sample will not be played.

"Hi vel" is the highest velocity value that should trigger the sample, which means that if a key is pressed so hard that the velocity is higher than this value, the sample will not be played.

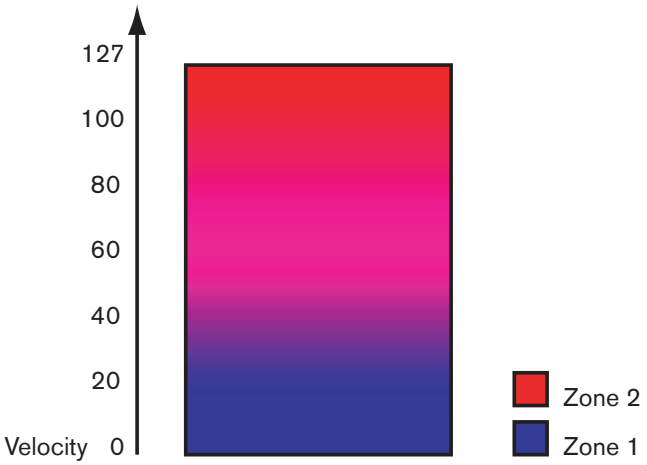
About Crossfading Between Zones

At the bottom right in the sample parameter area are two knobs marked "Fade In" and "Fade Out". These are primarily used for setting up velocity crossfades for smooth transitions between overlapping zones. In order to set up crossfades you adjust the fade out and fade in values for the overlapping zones.

Crossfading Between two Sounds

An example:

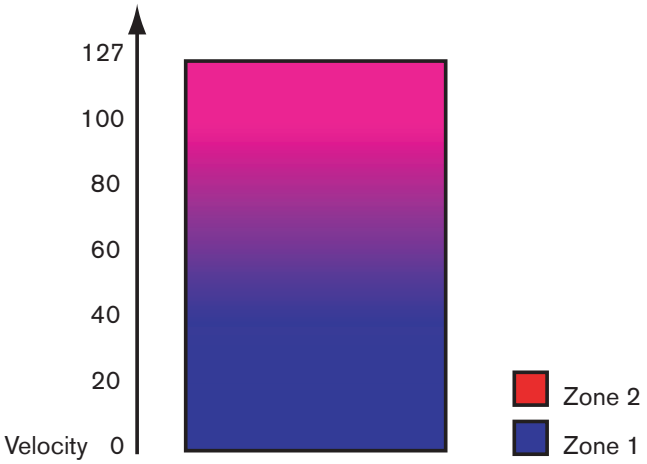
- **Two zones are both set to play in the full velocity range of 1-127.**
- **Zone 1 has a fade out value of 40.**
 This means that this zone will play at full level with velocity values below 40, With higher velocity values, it will gradually fade out.
- **Zone 2 has a fade in value of 80.**
 This has the effect that as you play velocity values up to 80, this zone will gradually fade in. With velocity values above 80, it will play at full level.



Another example:

Crossfading can be used to only fade in or fade out a certain sound. One common example is to set things up so that one sound plays the entire velocity range and another is faded in only at high velocity values.

- **Zone 1 is set to play the entire velocity range with no crossfade.**
- **Zone 2 is set to play the velocity range 80 to 127, with a fade in value of 110.**
 This means that this zone will start fading in from velocity values 80 and will play at full level in the velocity range 110 to 127.



This can be used for example to add a rimshot to a regular snare sound or a harder attack to a softer violin sample.

Setting crossfading for a Zone

Manually

To set up a crossfade for a zone, proceed as follows:

1. **Select one or more zones that you want to adjust.**
2. **Use the knobs marked “Fade In” and “Fade Out” in the sample parameter area, to set the desired values.**



- **You can change the values with finer precision by pressing [Shift] while turning the knobs, and you can reset the standard values by pressing [Command] (Mac)/[Ctrl] (Windows) and clicking on the knobs.**

Automatically

If you find it tedious to manually set up crossfades between zones, NN-XT can do it for you! The Edit menu and the NN-XT context menu contain an item called “Create Velocity Crossfades”.

1. **Set up the zones so that their velocity ranges overlap, as desired.**
2. **Select the zones.**
 You can select as many zones as you wish, not just one pair of overlapping zones.
3. **Select “Create Velocity Crossfades” from the Edit menu.**
 NN-XT will analyze the overlapping zones and automatically set up what it deems to be appropriate fade in and fade out values for the zones.

→ **This operation will not work if both zones have full velocity ranges.**
At least one of the zones must have a partial velocity range (see “About full and partial velocity ranges”).

→ **This operation will not work if the zones are completely overlapping.**

Using Alternate

About the Alternate function



At the bottom right in the sample parameters area is a knob marked “Alt”. It only has two states - On and Off. This is used for semi-randomly alternating between zones during playback.

There are several practical uses for this. Here follows two examples:

- **Layering several recordings of the same snare drum. By alternating between them you get a more natural repetition.**
- **Layering string up- and down strokes. By alternating you get the realistic effect of switching between the two directions of the stroke.**

You can layer as many sounds as you will and the algorithm switches between them in a way that provides as little repetition as possible.

To set up an alternating set of zones, proceed as follows:

1. **Set up the zones so that they overlap completely or partially.**
2. **Select them all.**
3. **Set "Alt" to On for all the zones.**
 Now, the program will automatically detect how to alternate between the zones, depending on their overlap.

Sample parameters



The Sample parameter area is found below the screen. They allow you to adjust parameters for one or several selected zones. Adjusting a parameter with multiple zones selected, will set the parameter to the same value for all selected zones. Below follows a run-down of the various parameters:

Root Note and Tune

These parameters are described in "Setting Root Notes and Tuning".

Sample Start and End

By turning the knobs you offset the start and end positions, so that they will play back more or less of a sample's waveform. Typical examples of use for this would be:

- **Removing unwanted portions from samples.**
 This could be anything from noise to "dead air" at the beginning or end of a sample.
- **To create variations out of a single sample.**
 These controls can be used to pick out any section of a recording for use as a sample.
- **Together with velocity sample start control.**
 You can for example increase Sample Start and then apply negative velocity modulation to Sample Start. Then, the harder you play the more you will hear of the attack portion of the sound.

► **If you hold down [Shift] when adjusting these parameters, the adjustment is in single frames (samples).**

Loop Start and End

A sample, unlike the cycles of an oscillator for example, is a finite quantity. There is a sample start and end. To get samples to play for as long as you press down the keys on your keyboard, they need to be looped. For this to work properly, you have to first set up two loop points which determine the part of the sample that will be looped.

The instrument samples in the sound banks included with Reason are already looped. The same will be true for most commercial sample libraries. However, if you need to, you can use these controls to adjust the looping.

- **The size and position of the loop – in the sample – is determined by two parameters, Loop Start (the beginning of the loop) and Loop End (the end point of the loop).**
- **The NN-XT then keeps repeating the section between the Loop Start and Loop end until the sound has decayed to silence.**

Play Mode

By using this knob you can select one of the following loop modes for each zone:

- **FW**
 The sample in the zone will play only once, without looping.
- **FW-LOOP**
 The sample will play from the sample start point to the loop end point, jump back to the loop start point and then loop infinitely between the start and end loop points. This is the most common loop mode.
- **FW - BW**
 The sample will play from the sample start point to the loop end point, then from the loop end point to the loop start point (backwards), and then loop infinitely forwards-backwards between the start and end loop points.
- **FW-SUS**
 This works like FW-LOOP with the exception that it will only loop as long as the key is held down. As you release the key, the sample will play to the absolute end of the sample, that is beyond the boundaries of the loop.

 This means that the sound may have a short natural release even if the release parameter is raised to a high value (which is not true for "FW-LOOP", where the release parameter always controls the length of the sound after the key is released).
- **BW**
 The sample will play only once - from the end to the beginning - without looping.

Lo Key and Hi Key

These parameters are described in "Setting up Key Ranges".

Lo Vel and Hi Vel

These parameters are described in "About velocity ranges".

Fade In and Fade Out

These parameters are described in "About Crossfading Between Zones".

Alt

This parameter is described in "About the Alternate function".

Out

The NN-XT features eight separate stereo output pairs (see "Audio Output"). For each zone, you can decide which of these output pairs to use. Thus, if you have created a key map consisting of eight zones, each of these can have a separate stereo output from NN-XT, and can then be routed to a separate mixer channel if you so wish.

→ **To select which output a selected zone should be directed to, use the knob marked "Out" in the sample parameter area.**
The output pairs are indicated above the button.

! **Note that you still have to route the outputs the way you want them on NN-XT's back panel. If you assign a zone to an output pair other than 1-2 (which is the default) no connections or auto routing are made. You have to do that manually.**

A Stereo example

One possible way of utilizing this would be to create a drum kit. In this case you could load up to eight different stereo drum samples, assign them to separate outputs, route each to a separate mixer channel and then use the mixer to set levels and pan, add send effects etc.

Using a stereo output as two mono outputs

If, on the other hand, you are using mono samples, you can use one stereo pair as a two separate outputs, effectively giving you a total of 16 separate outputs.

1. **Assign two zones to the same output.**
2. **Us the Pan control to pan one of the zones hard left and the other hard right.**
3. **Connect each of the two outputs in the stereo pair to a separate mixer channel.**

Group parameters



The group parameters are located at the top left on the remote editor panel. These are parameters that in various ways are directly related to playing style.

Group parameters apply to a group, that is they are settings that are shared by all zones in a group.

- **To make adjustments to one group, select one or more zones that belong to the group, and adjust the parameter on the front panel.**
- **To set several groups to the same value, select at least one zone in each group you want to adjust, and adjust the parameter on the front panel.**

Key Poly

This setting determines the number of keys that you can play simultaneously (the polyphony). The maximum number is 99 and the minimum is 1, in which case the group will be monophonic.

Users of other samplers may want to note that the polyphony often means setting the number of *voices* that should be able to play. The NN-XT is different in this aspect, since the polyphony setting instead determines the number of *keys,* regardless of how many voices each key plays.

The Group Mono button

The Group Mono button beside the Key Poly section can be used to quickly set a group to play monophonically regardless of the polyphony setting. E.g. if you have a group with open and closed hi hats, you can switch this on so that an open hi hat is automatically muted when you play a closed hi hat.

Group Mono overrides the Key Poly setting - except when playing the same note.

So you can play your open hi-hat repeatedly without the sound cutting itself off. When you play the closed hi-hat, this cuts off the open hi-hat.

Note that activating this button is not the same as setting polyphony to 1. E.g., it can not be used for Legato or mono Retrig (see "Legato and Retrig").

Legato and Retrig

Legato

Legato works best with monophonic sounds. Set Key Poly (see above) to 1 and try the following:

- **Hold down a key and then press another key without releasing the previous.**
 Notice that the pitch changes, but the envelopes do not start over. That is, there will be no new "attack".
- **If Key Poly is set to more voices than 1, Legato will only be applied when all the assigned keys are "used up".**
 For example, if you had a polyphony setting of "4" and you held down a 4 note chord, the next note you played would be Legato. Note, however, that this Legato key will "steal" one of the keys in the 4 note chord, as all the assigned keys were already used up!

Retrig

Retrig is the "normal" setting for playing polyphonic patches. That is, when you press a key without releasing the previous, the envelopes are triggered, like when you release all keys and then press a new one. In monophonic mode, Retrig has an additional function; if you press a key, hold it, press a new key and then release that, the first note is also retriggered.

LFO 1 Rate

This is used for controlling the rate of LFO 1 if it is used in "Group Rate" mode. In that case, this knob will take precedence over the rate parameter in the LFO 1 section. See "The LFOs" for detailed information about this.

Portamento

This is used for controlling portamento - a parameter that makes the pitch glide between the notes you play, rather than changing the pitch instantly as soon as you hit a key on your keyboard. By turning this knob you set how long it should take for the pitch to glide from one note to the next as you play them.

In legato mode, there will only be any portamento when actually playing legato (tied) notes.

With the knob turned all the way to the left, portamento is disabled.

Synth parameters

The Modulation controls



As previously described, the Modulation wheel (and the External Control wheel) can be used for controlling various parameters. These controls allow you to define which parameters the wheels should modulate and to what extent.

- **Below each of the knobs are the letters "W" and "X".**
 These are used for selecting the source that should control the parameter, and represent the "Modulation Wheel" and the "External Control wheel" respectively.
- **By clicking on any of the letters, you decide which source should control the parameter.**
 You can select either, both or none. When a letter is "lit", the corresponding source is set to control the parameter.
- **By turning the knobs, you decide how *much* the modulation and/or external control wheel should modulate the corresponding parameter.**

Note that all of the control knobs are bi-polar, which means that they can be set to both positive and negative values. Positive values are set by turning the knobs to the right, and negative values are thus set by turning the knobs to the left:

- **Setting them to positive values means that the value of the controlled parameter will be raised if the source wheel is pushed forward.**
- **Setting them to negative values means that the value will be lowered when a wheel is pushed forward.**
- **Keeping the knobs in the center position means that no modulation control is applied.**

There is one exception to these rules, and that is the LFO 1 Amt control, which works in a slightly different way. See below for more information about this.

The following parameters can be modulated:

F.Freq

This sets modulation control of the Filter's cutoff frequency (see "The Filter section").

Mod Dec

This sets modulation control of the Decay parameter in the Modulation Envelope (see "The Modulation Envelope").

LFO 1 Amt

This determines how much the amount of modulation from LFO 1 is affected by the Modulation wheel and/or the External Controller wheel. It does this by "scaling" the amounts set with the three destination knobs in the LFO 1 section (Pitch, Filter and Level, see "The LFOs"). We'll explain this with an example:

To use the Modulation Wheel to *increase* pitch modulation (vibrato), proceed as follows:

1. **Turn the Mod Wheel all the way down, so that no modulation is applied.**
2. **Activate the "W" button for LFO 1 Amt in the Modulation section.**
3. **Set the corresponding knob to "12 o'clock" (zero).**
4. **Set up LFO 1 so that as much vibrato is applied as you want it to be when the Modulation wheel is turned all the way up.**
5. **Increase LFO 1 Amt until you hear as much vibrato as you want it to be when the wheel is turned all the way down.**
 If you turn LFO 1 Amt all the way up, there will be no vibrato at all when the wheel is all the way down.

To instead use the Modulation wheel to *decrease* vibrato, process as follows:

1. **Turn the Mod Wheel all the way down, so that no modulation is applied.**
2. **Activate the "W" button for LFO 1 Amt in the Modulation section.**
3. **Set the corresponding knob to "12 o'clock" (zero).**
4. **Set up LFO 1 so that as much vibrato is applied as you want it to be when the Modulation wheel is turned all the way down.**
5. **Turn the Modulation wheel all the way up.**
6. **Decrease LFO 1 Amt until you hear as much vibrato as you want it to be when the wheel is turned all the way up.**
 If you turn LFO 1 Amt all the way down, there will be no vibrato at all when the wheel is all the way up.

F.Res

This sets modulation control of the Resonance parameter in the Filter (see "The Filter section").

Level

This sets the amount of amplitude envelope modulation of each zone's level. The level set here will be the level of the highest point of the Amp Envelope.

LFO 1 Rate

This sets modulation control of the Rate parameter in LFO 1 (see "The LFOs").

The Velocity controls



Velocity is used for controlling various parameters according to how hard or soft you play notes on your keyboard. A typical use of velocity control is to make sounds brighter and louder if you strike a key harder. By using the knobs in this section, you can control if and how much the various parameters will be affected by velocity.

Just like the modulation controls, all of the velocity control knobs are bi-polar, and can be set to both positive and negative values.

- **Setting them to positive values means that the value of the controlled parameter will be raised the harder you play.**
- **Setting them to negative values means that the value will be lowered the harder you play.**
- **Keeping the knobs in the center position means that no velocity control is applied.**

The following parameters can be velocity controlled:

F.Freq

This sets velocity control of the Filter's cutoff frequency (see "The Filter section").

Mod Dec

This sets velocity control of the Decay parameter in the Modulation Envelope (see "The Modulation Envelope").

Level

This sets velocity control of the Amp Envelope.

Amp Env Attack

This sets velocity control of the Attack parameter in the Amplitude Envelope (see "The Amplitude Envelope").

Sample Start

This sets velocity control of the Sample Start parameter (see "Sample Start and End"), so that it will be offset forwards or backwards, according to how hard or soft you play.

This allows you to control how much of the attack portion of the sample you hear when playing harder or softer.

To be able to make use of negative values for this parameter, you must increase the sample parameter Sample Start.

The Pitch section



This section contains various parameters related to controlling the pitch, or frequency, of the zones.

Pitch Bend Range

This lets you set the amount of pitch bend, i.e. how much the pitch changes when your turn the pitch bend wheel fully up or down. The maximum range is +/- 24 semitones (2 Octaves).

Setting the pitch

Use the three knobs marked “Octave”, “Semi” and “Fine” to change the pitch of the sample(s):

- **Octave**
 This changes the pitch in steps of one full octave. The range is -5 – 0 – 5.
- **Semi**
 This lets you change the pitch in semitone steps. The range is -12 – 0 – 12 (2 octaves).
- **Fine**
 This changes the pitch in cents (hundredths of a semitone). The range is -50 – 0 – 50 (down or up half a semitone).

K. Track

This knob controls Keyboard Tracking of the pitch.

- **In the center position, each key represents a semitone This is the normal setting.**
- **When turned all the way down, all keys play the same pitch. This can be useful for percussion like timpani where you might want to play the same pitch from a range of keys.**
- **When turned all the way up, each key on the keyboard shifts the pitch one octave.**

The Filter section



Filters can be used for shaping the character of the sound. The filter in NN-XT is a multimode filter with six different filter types.

→ **To activate/deactivate the filter, click the On/Off button in the top right corner.**
When the filter is activated, the button is lit.

Filter mode

To select a filter mode, either click the Mode button in the bottom right corner or click directly on the desired filter name so that it lights up:

- **Notch**
 The notch filter is used for cutting off frequencies in a narrow frequency range around the set cutoff frequency, while letting the frequencies below and above through.
- **HP 12**
 This is a highpass filter with a 12 dB/Octave roll-off slope. A highpass filter cuts off low frequencies and lets high frequencies pass. That is, frequencies below the cutoff frequency are cut off and frequencies above it pass through.
- **BP 12**
 This is a bandpass filter with a 12 dB/Octave roll-off slope. A bandpass filter could be viewed as the opposite of a notch filter. It cuts off both the high and the low frequencies, while frequencies in the band range pass through.
- **LP 6**
 This is a lowpass filter with a gentle, 6 dB/Octave slope. A lowpass filter is the opposite of a highpass filter. It lets the low frequencies through and filters out the high frequencies. This filter has no Resonance.
- **LP 12**
 This is a lowpass filter with a 12 dB/Octave roll-off slope.
- **LP 24**
 This is a lowpass filter with a fairly steep roll-off slope of 24 dB/Octave.

Filter controls

The following filter controls are available:

- **Freq**
 This is used for setting the filter cutoff frequency. The cutoff frequency determines the limit above or below which frequencies will be cut off depending on the selected filter type. In the case of a lowpass filter for example, frequencies below the cutoff frequency will be allowed to pass through, while frequencies above it will be cut off. The farther to the right you turn the knob, the higher the cutoff frequency will be.

 ► **It is very common to modulate filter frequency with the modulation envelope, as described in "The Modulation Envelope".**

- **Res**
 Technically, this knob controls feedback of the output signal from the filter, back to its input. Acoustically it emphasizes frequencies around the cutoff frequency. For a lowpass filter for example, increasing Res will make the sound increasingly more hollow until the sound starts "ringing". If you set a high value for the Res parameter and then vary the filter frequency, this will produce a classic synthesizer filter sweep.

 For the notch and bandpass filter types, the Resonance setting instead adjusts the width of the band. That is, the higher the resonance setting, the narrower the band will be where frequencies are cut off (notch) or let through (Bandpass).

- **K. Track**
 This lets you activate and control keyboard tracking of the filter frequency. If keyboard tracking is activated, the set cutoff frequency of the filter will change according to the notes you play on your keyboard. That is, if you play notes higher up on the keyboard, the filter frequency will be raised and vice versa.

 When the knob is set to its center position, filter frequency is adjusted so that the harmonic content remains constant across the keyboard.

 Keyboard tracking is deactivated by default (the knob all the way to the left). This means that the filter frequency will remain unchanged regardless of where on the keyboard you play.

The Modulation Envelope



The Modulation Envelope parameters let you control how certain parameters, or destinations, should change over time - from the moment a note is struck to the moment it is released again.

The destinations you can use are:

- **Pitch**
- **Filter frequency**

Parameters

The following are the available controlling parameters:

- **Attack**
 When you press a key on your keyboard, the envelope is triggered. The attack parameter then controls how long it should take before the controlled parameter (pitch or filter) reaches the maximum value, when you press a key. By setting attack to a value of "0", the destination parameter would reach the maximum value instantly. By raising the attack parameter, the value will instead slowly "slide" up to its maximum.
- **Hold**
 This is used for deciding how long the controlled parameter should stay at its maximum value before starting to decrease again. This can be used in combination with the Attack and Decay parameters to make a value reach its maximum level, stay there for a while (hold) and then start dropping gradually down to the sustain level.
- **Decay**
 After the maximum value for a destination has been reached and the Hold time has expired, the controlled parameter will start to gradually drop down to the sustain level. How long it should take before it reaches the sustain level is controlled with the Decay parameter. If Decay is set to "0", the value will immediately drop down to the sustain level.
- **Sustain**
 The Sustain parameter determines the value the envelope should drop back to after the Decay. If you set Sustain to full level however, the Decay setting doesn't matter since the value will never decrease.

 A combination of Decay and Sustain can be used for creating envelopes that rise up to the maximum value, then gradually decrease to, and stay on a level somewhere in-between zero and maximum.
- **Release**
 This works just like the Decay parameter, with the exception that it determines the time it takes for the value to fall back to zero *after* the key is released.
- **Delay**
 This is used for setting a delay between when a note is played and when the effect of the envelope starts. That is, the sound will start unmodulated, and the envelope will kick in after you have kept the key(s) pressed down for a while. Turn the knob to the right to increase the delay time. If the knob is set all the way to the left, there will be no delay.
- **Key To Decay**
 By using this, you can cause the value of the Decay parameter (see above) to be offset depending on where on your keyboard you play. If you turn the knob to the right the decay value will be raised the higher up you play, and turning the knob to the left will lower the decay value the higher up you play. With the knob in the center position, this parameter is deactivated.

Destinations

The following are the available Mod Envelope destinations:

- **Pitch**
 This will make the envelope modulate the pitch, as set in the Pitch section (see "The Pitch section"). Turn the knob to the right to raise the pitch and to the left to lower the pitch. In the middle position, pitch will not be affected by the envelope.
- **Filter**
 This will make the envelope modulate the cutoff frequency of the Filter (see "The Filter section"). Turn the knob to the right to increase the frequency and to the left to lower the frequency. In the middle position, the envelope will have no effect on the cutoff frequency.

The Amplitude Envelope



The Amplitude Envelope parameters let you control how the volume of a sound should change over time - from the moment a note is struck to the moment it is released again.

Parameters

Most of the Amplitude Envelope parameters are identical to those of the Modulation Envelope. So for a detailed description of the following parameters, please refer to the modulation envelope section in "The Modulation Envelope":

- **Attack**
- **Hold**
- **Decay**
- **Sustain**
- **Release**
- **Delay**
- **Key To Decay**

The following are the parameters that are unique for the Amp Envelope section:

- **Level**
 This knob sets the level of the zone. Turn it to the right to raise the level.
- **Spread and Pan modes**
 These two parameters are used for controlling the stereo (pan) position of the sound. The Spread knob determines the sound's width in the stereo image (how far left – right the notes will be spread out). If this is set to "0", no spread will take place. The Mode selector switch is used for choosing which type of spread you want to apply:

Mode	Description
Key	This will make the pan position shift gradually from left to right, the higher up on the keyboard you play.
Key 2	This will make the pan position shift from left to right and then back again from right to left in a sequence of eight keys. Playing 4 adjacent semitones thus makes the pan position gradually go from left to right. The next 4 higher semitone notes will then change the pan position from right to left in the same way, and this cycle will then be repeated.
Jump	This will make the pan position jump between left and right each time a note is played.

- **Pan**
 This controls the stereo balance of the output pair to which a zone is routed. In the middle position, the signal appears equally strong on the left and right channel in a stereo pair. By turning the knob to the left or right, you can change the stereo balance.

Note that if you for instance turn the Pan knob all the way to the left, you cause the signal to be output from the left channel of the stereo pair only.

You can use this to treat a stereo output as two independent mono outputs, if required.

See "Out" for information on routing zones to output pairs.

The LFOs



NN-XT features two Low Frequency Oscillators - LFO 1 and LFO 2. "Normal" oscillators generate a waveform and a frequency, and produce sound. *Low frequency* Oscillators on the other hand, also generate a waveform and a frequency, but there are two major differences:

- **LFOs only generate sounds of a low frequency.**
- **LFOs don't produce sound, but are instead used for modulating various parameters.**

The most typical use of an LFO is to modulate the pitch of a sound (generated by an oscillator or - in the case of NN-XT - a sample), to produce vibrato.

About the Difference between LFO 1 and LFO 2

There are two fundamental differences between LFO 1 and LFO 2:

- **LFO 2 is always key synced, that is, each time you press a key, the LFO waveform starts over from scratch. LFO 1 can be switched between key synced and non-key synced modes.**
- **LFO 2 only has one waveform, triangle.**

The following parameters are available for the LFOs:

Rate (LFO 1 and 2)

This knob controls the frequency of the LFO. For a faster modulation rate, turn the knob to the right. The Rate knob of LFO 1 is also used for setting the timedivision when synchronizing the LFO to the song tempo (see below).

Delay (LFO 1 and 2)

This can be used for setting a delay between when a note is played and when the LFO modulation starts kicking in (gradually). This way, you can make the sound start unmodulated, and then have the LFO modulation start after you have kept the key(s) pressed down for a while.

Turn the knob to the right to increase the delay time.

Mode (LFO 1 only)

This lets you set the "operation mode" for the LFO. Click the button to switch between the available modes:

- **Group Rate**
 In this mode, the LFO will run at the rate set for its group in the group section, rather than at the rate set here (see "Group parameters"). This way, all zones in the group will have the exact same modulation rate.
- **Tempo Sync**
 In this mode, the LFO will be synchronized to the song tempo, in one of 16 possible time divisions.

! **When tempo sync is activated, the Rate knob is used for selecting the desired timedivision. Turn the Rate knob and observe the tool tip for an indication of the timedivision.**

- **Free Run**
 In free run mode, the LFO simply runs at the rate set with the Rate parameter. Furthermore, if Key Sync is deactivated, the modulation cycle will not be retriggered each time you press a key - it will run continuously.

Waveform (LFO 1 only)

Here, you select which type of waveform should be used for modulating the destination parameters.

Click the button to switch between the following waveforms:

Waveform	Description
Triangle	This is a smooth waveform, suitable for normal vibrato.
Inverted Sawtooth	This produces a "ramp up" cycle. If applied to an oscillator's frequency, the pitch would sweep up, after which the cycle immediately starts over.
Sawtooth	This produces a "ramp down" cycle, the same as above but inverted.
Square	This produces cycles that abruptly change between two values, usable for trills etc.
Random	Produces random stepped modulation to the destination. Some vintage analog synths called this feature "sample & hold".
Soft Random	The same as above, but with smooth modulation.

! **LFO 2 always uses a triangle waveform.**

Key Sync (LFO 1 only)

By activating key sync, you "force" the LFO to restart its modulation cycle each time a key is pressed.

! **Note that LFO 2 always uses Key Sync.**

Destinations for LFO 1

The following parameters can be modulated by LFO 1:

- **Pitch**
 This will make the LFO modulate the pitch, for vibrato, trills, etc. It can be set to -2400 – 0 – 2400 cents which equals 4 octaves. The set pitch will change up and down by this amount, with each modulation cycle. Turning the knob to the right will make the modulation cycle start above the set pitch, while turning it to the left will invert the cycle. Keeping this in the middle position means that the pitch will not be affected by the LFO.
- **Filter**
 This will make the LFO modulate the cutoff frequency of the Filter, for auto-wah effects, etc. The positive/negative effect is the same as for pitch.
- **Level**
 This will make the LFO modulate NN-XT's output level, for tremolo effects, etc. The positive/negative effect is the same as for pitch.

Destinations for LFO 2

The following parameters can be modulated by LFO 2:

- **Pan**
 This makes the LFO modulate the pan position of a zone. The sound will move back and forth in the stereo field. Turning the knob to the left makes the sound move from left to right, and turning it to the right thus makes it move from right to left. The middle position provides no modulation at all.
- **Pitch**
 Just like for LFO 1 (see above), this makes LFO 2 modulate the pitch. The range is also the same as for LFO 1.

Connections

On the back panel of NN-XT are a number of connectors. Many of these are CV/Gate related. Using CV/Gate is described in the chapter “Routing Audio and CV”.



Sequencer Control

The Sequencer Control CV and Gate inputs allow you to play the NN-XT from another CV/Gate device (typically a Matrix or a Redrum). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity.

Modulation Input

These control voltage (CV) inputs (with associated voltage trim pots), can modulate various NN-XT parameters from other devices. These inputs can control the following parameters:

- **Oscillator Pitch**
- **Filter Cutoff Frequency**
- **Filter Resonance**
- **LFO 1 Rate**
- **Master Volume**
- **Pan**
- **Modulation Wheel**
- **Pitch Wheel**

Gate Input

These inputs can receive a CV signal to trigger the following envelopes:

- **Amplitude Envelope**
- **Modulation Envelope**

Note that connecting to these inputs will override the normal triggering of the envelopes. For example, if you connect a Matrix Gate Out to the Gate In Amp Envelope, you would not trigger the amp envelope by playing notes, as this is now controlled by the Matrix Gate Out. In addition you would only hear the Gate Out triggering the envelope for the notes that you hold down.

Audio Output

There are 16 audio output jacks on the NN-XT's back panel - eight separate *stereo pairs*. When you create a new NN-XT device, the first output pair (1L & 2R) is auto-routed to the first available outputs in the I/O device.

The other output pairs are never automatically routed. If you wish to use any of the other output pairs, you have to manually connect them to the desired device. The basics on Routing is described in “Routing Audio and CV”.

! **Note that when you use any other output pair than the first, you also have to route one or more zones to it if you want it to actually output sound, since all zones by default are routed to outputs 1 & 2. How to route zones to other outputs is described in the "Out" section.**

Chapter 27
NN-19 Sampler

Introduction

The NN-19 is a sampler capable of playing back - but not editing - sound files.

The program comes with numerous ready-made sample patches, covering all kinds of instrument types. In addition to this there are plenty of single samples that can be used for creating your own patches.

There are also plenty of relatively inexpensive (and even free) audio editing software for both the Windows and the macOS platforms, that will allow you to both record audio (via your computers or audio cards audio inputs), and to edit the resulting audio file. Virtually every software that is capable of this, can create sound files which can be loaded directly into the NN-19.

Also, there are thousands of high quality sample libraries available, covering every conceivable musical style or direction ranging from professionally recorded orchestral samples to esoteric electronic noises.

! Please, note that this device is not available in Reason Lite Rack Plugin.

General sampling principles

Background

Before a sound can be used by a sampler, it must be converted to a digital signal. Hardware samplers and computer audio cards provide audio inputs that can convert the analog signal to digital, by the use of an "A/D Converter" (analog to digital). This "samples" the signal at very short time intervals and converts it to a digital representation of the analog signal's waveform. The sample rate and the bit depth of this conversion determines the resulting sound quality. Finally the signal is passed through a digital to analog converter (D/A) which reconstructs the digital signal back to analog, which can be played back.

Multisampling vs. single samples

Most of the included NN-19 patches are made up of a collection of several samples. This is because a single sampled sound only sounds natural within a fairly narrow frequency range. If a single sample is loaded into an empty NN-19, the sample will be playable across the whole keyboard. The pitch (frequency) of the original sample (called root-key) will be automatically placed on the middle C key (C3).

Note that this has nothing to do with the actual pitch the sample itself produces! It may not even have a pitch as such, it could be the sound of someone talking for example.

If you play any single sample about two octaves above or below its root key, it will most likely sound very "unnatural". In the case of it actually being a sample of someone talking, playing two octaves up will make the talking voice sample sound squeaky, short and most likely unintelligible. Two octaves down the voice will sound something like a drawn-out gargle.

Thus, the range that most samples can be transposed without sounding unnatural is limited. To make a sampled piano, for example, sound good across the whole keyboard, you need to first have made many samples at close intervals across the keyboard, and then define an upper and lower range for each sample, called a Key Zone. All the keyzones in the piano sample patch then make up a Key Map.

How to create key zones is described in "About Key Zones and samples".

To sample real instruments accurately requires a lot of hard work. Firstly, you need the original instrument, which should be in perfect working order. For acoustic instruments you need a couple of good microphones, a mixer or other device with high quality microphone preamps, and a room with good acoustics. You need to be meticulous when recording the different samples, so that levels are smooth and even across the range etc.

Fortunately Reason provides a wide range of high quality multisampled instruments, so much of this hard work has already been done for you.

In our experience, most people don't use samplers only for playing sampled versions of "real" instruments. Very often, single "stand alone" or single samples are used. Maybe you wish to use different sounds for every key zone. Or you could have complete chorus and verse vocals plus variations assigned to several "one note" key zones. Or use samples of different chords that play rhythmic figures to the same tempo, and use these to build song structures etc. The possibilities are endless. When you use samples in this way, the keys on your keyboard that play the samples do not necessarily correspond to pitch at all, the keys are simply used to trigger the samples.

About audio file formats

The audio file format support differs depending on which computer OS you are using.

The NN-19 can read audio files in the following formats:

- **In Windows:**
 .wav, .aif, .mp3, .aac, .m4a and .wma.
- **In macOS:**
 .wav, .aiff, .3g2, .3gp, .mp1, .mp2, .mp3, .mpeg, .mpa, .snd, .au, .sd2, .ac3, .aac, .adts, .amr, .caf, .m4a .m4r and .mp4.
- **SoundFonts (.sf2)**
 SoundFonts are an open standard for wavetable synthesized audio, developed by E-mu systems and Creative Technologies.
- **REX file slices (.rx2, .rex, .rcy)**
 REX files are music loops created in the ReCycle program. The NN-19 lets you either load REX files as patches or separate slices from REX files as individual samples.
- **Any sample rate and practically any bit depth.**

About the Sample Patch format

Reason's Sample Patch format (.smp), is based on either Wave or AIFF files, but includes all the NN-19 associated parameter settings as well.

- **The audio files may be stereo or mono. Stereo audio files are shown with a "S" symbol beside its name in the display.**

Loading a Sample Patch

When you create a new NN-19 device, it is automatically loaded with a default patch. If you want to start from scratch, with no samples loaded, you can select "Reset Device" from the context menu or Edit menu. For NN-19 to produce sound, you need to load either a sample patch, or a sample.

A patch contains "everything". All the samples, assigned key zones, and associated panel settings will be loaded. Loading a sample patch is done using the Browser, just like in all other devices that use Patches.

1. **Click the Browse Patch button on the front panel to set browse focus to the NN-19 device.**
2. **Navigate to the folder that contains the NN-19 patch you wish to load, select it and click Load in the Browser.**

→ **Alternatively, drag an NN-19 patch from the Browser and drop it on the NN-19 device in the rack.**
The panel is dimmed in orange and the Patch Replace symbol appears in the center.

Loading REX Files as Patches

REX files are music loops created in the ReCycle program. In Reason, REX files are primarily used in the Dr. Octo Rex loop player, but they can be used in the NN-19 as well. Possible extensions are ".rx2", ".rcy" and ".rex".

1. **Click the Browse Patch button on the front panel to set browse focus to the NN-19 device.**
2. **Navigate to the folder that contains the REX loop you wish to load, select it and click Load in the Browser.**

→ **Alternatively, drag a REX loop from the Browser and drop it on the NN-19 device in the rack.**
The panel is dimmed in orange and the Patch Replace symbol appears in the center.

When loading a REX file, each slice in the file is assigned to one key, chromatically, starting from C1. All parameters are set to their default settings.

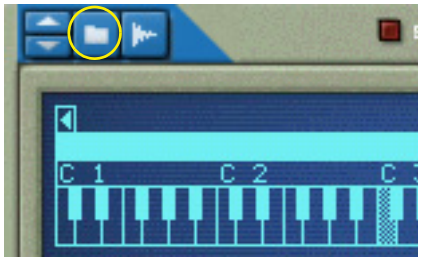
When using REX files in the Dr. Octo Rex loop player, it is possible to make a track play the slices in order to recreate the original loop. To do the same in the NN-19 requires a few extra steps.

1. **Use the Browser to load the REX file into an NN-19 sampler.**
2. **Create a Dr. Octo Rex loop player and load the same REX file into o Loop Slot of this device.**
3. **Use the "Copy Loop To Track" feature on the Dr. Octo Rex to create playback data (a group) on the track assigned to the Dr. Octo Rex.**
4. **Move that group to the track that plays the NN-19 and play it back from there.**
5. **Delete the Dr. Octo Rex loop player.**

About Key Zones and samples

Loading a Sample into an empty NN-19

1. **Create an NN-19 device and select "Reset Device" from the context menu or from the Edit menu.**
2. **Click on the Browse Sample button.**
This is located above the keyboard display to the left.



← **When you browse samples, you can preview them before loading using the Play button in the Browser. If you select the "Autoplay" function, the samples play back once automatically when selected.**

3. **Select a sample in the Browser and click the Load button in the Browser to load it.**

→ **Alternatively, drag a sample file from the Browser and drop it on the NN-19 device in the rack.**
The panel is dimmed in blue and a Sample Replace symbol appears in the center.

When you load the first sample into an empty NN-19, this will be assigned a key zone that spans the entire range of the keyboard, and the default Init Patch settings will be used.

Below the keyboard, the range, sample name, root key, tuning, level and loop status of the current key zone is displayed, each with a corresponding knob.



The light blue strip above the keyboard indicates the currently selected key zone, which is in this case the full range of the keyboard.

The inverted note on the keyboard indicates the "root key" of the sample. All samples contain a root key, tuning and level setting. If NN-19 is empty, a sample will have its root key placed on the middle "C" (C3) key.

4. **If desired, click on the keyboard to change the root key.**

! **You can audition a loaded sample patch or sample by holding down [Option] (Mac)/[Alt] (Windows) and clicking on a key in the Keyboard display. The mouse will take on the shape of a speaker symbol to indicate this.**

Loading SoundFont samples

The SoundFont format was developed by E-mu systems in collaboration with Creative Technologies. It is a standardized data format containing wavetable synthesized audio and information on how it should be played back in wavetable synthesizers - typically on audio cards. The SoundFont format is an open standard so there is a vast amount of SoundFont banks and SoundFont compatible banks developed by third parties.

The samples in a SoundFont are stored hierarchically in different categories: User Samples, Instruments, Presets etc. The NN-19 allows you to browse for and load single SoundFont samples, but *not* entire soundfonts.

1. **Click the Browse Sample button, select a SoundFont file (.sf2) in the Browser and open it.**
 The Browser opens the SoundFont and displays the folders within it.
2. **Select the folder "Samples" and open it.**
 This folder contains a number of samples which can be loaded like any other sample.
3. **Select the desired sample and load it by clicking the Load button in the Browser.**
 The sample is loaded and assigned a key zone range that spans the entire keyboard. You can now make settings for it as with any other sample.

→ **Alternatively, drag a sample file from the Browser and drop it on the NN-19 device in the rack.**
The panel is dimmed in blue and a Sample Replace symbol appears in the center.

Loading REX slices as samples

A slice is a snippet of sound in a REX File.

1. **To import a REX slice, click the Browse Sample button (see above), browse to a REX file and open it as if it was a folder.**
 The Browser will then display the slices as files inside that "folder".

2. **Select the desired REX slice and load it by clicking the Load button in the Browser.**
 The REX slice is loaded and assigned a key zone range that spans the entire keyboard. You can now make settings for it as with any other sample.

→ **Alternatively, drag REX slice from the Browser and drop it on the NN-19 device in the rack.**
 The panel is dimmed in blue and a Sample Replace symbol appears in the center.

In the rest of this manual, when we refer to importing samples, all that is said applies to REX slices as well.

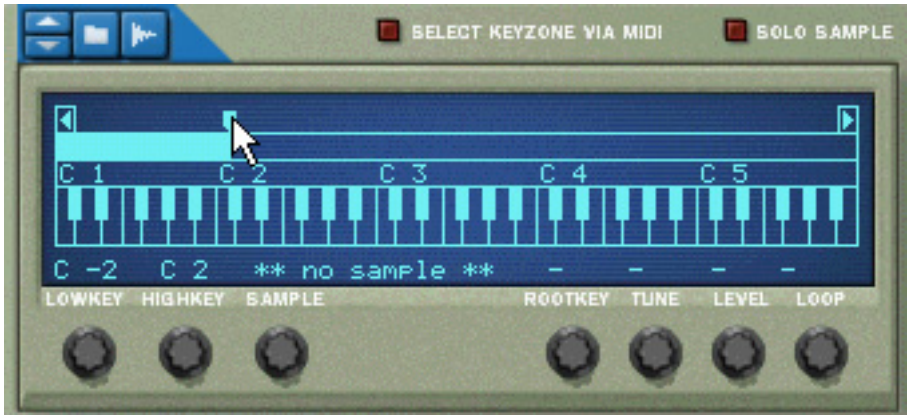
Creating Key Zones

A "key zone" is a range of keys, that plays a sample. All key zones together make up a "key map".

To create a new key zone, the following methods can be used:

→ **Select "Split Key Zone" from the Edit or context menus.**
 This splits the currently selected key zone in the middle. The new zone is the upper half of the split, and is empty. The dividing point has a "handle" above it, see "Setting the Key Zone Range" below for a description.

→ **By [Alt]/[Option]-clicking at a point just above the key zone strip, a new empty key zone is created.**
 The point where you click becomes the lower limit (or boundary) for the original key zone, and the upper limit for the new key zone.



The new empty key zone gets selected upon creation.

Selecting Key Zones

Only one key zone can be selected at a time. A selected key zone is indicated by a light blue (as opposed to dark blue) strip above the keyboard in the display. There are two ways you can select key zones:

→ **By clicking on a non-selected key zone in the display.**

→ **By activating the "Select Key Zone via MIDI" button.**
 Playing a note belonging to a non-selected key zone from your MIDI keyboard, will select the key zone it belongs to.

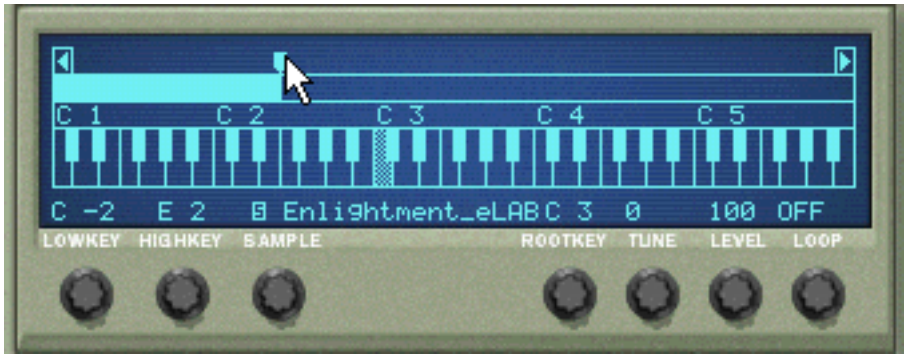
Setting the Key Zone Range

! **Key zones cannot overlap.**

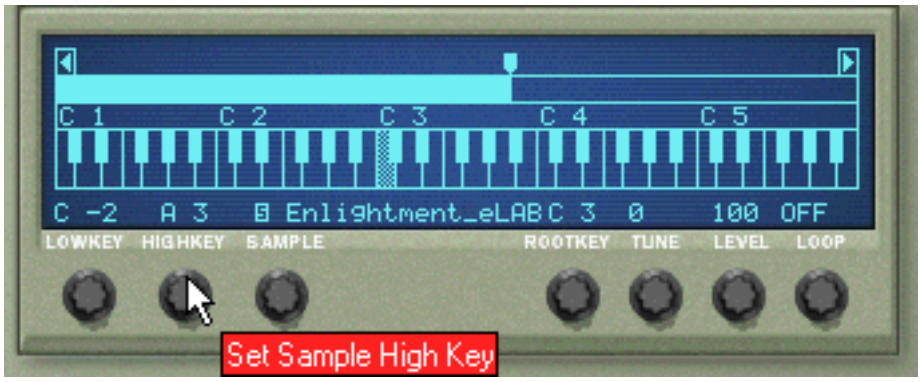
When you adjust the boundaries of a key zone, the surrounding boundaries are automatically adjusted accordingly.

You can change the key zone range in the following ways:

→ **By dragging the "handle(s)" which divides the key zones, you can change the range of the selected key zone.** In the case of having two key zones split in the middle, you could thus change the lower limit for the upper (new) key zone and the upper limit for the original key zone.



→ **By using the "Lowkey" and "Highkey" knobs to set a lower and upper range, respectively.**



Deleting a Key Zone

→ **To delete a key zone, select it and then select "Delete Key Zone" from the Edit menu.**

About Key zones, assigned and unassigned samples

When you load samples and rearrange your key mapping, you will often end up with samples that are not assigned to any key zone. In the following texts we refer to the samples as follows:

- **Assigned samples are samples that are currently assigned to one ore more key zones.**
- **Unassigned samples are samples that reside in the sample memory, but that are currently not assigned to any key zone.**

Adding sample(s) to a Key Map

If the sample hasn't been loaded yet

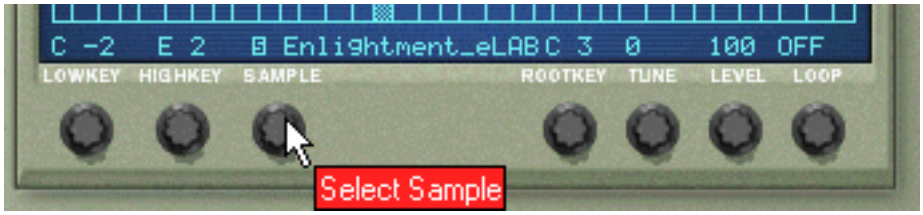
1. **Select a key zone.**
 This can be empty, or contain a sample - it doesn't matter for now.
2. **Use the Browser to add one, or several (see below), sample(s).**

The following will happen:

- **If the zone contained a sample prior to loading, this will be *replaced*, both in the zone and in the sample memory, unless the sample was also used by *another* key zone, in which case it will be kept.**
- **If you loaded several samples, one of the samples will be assigned to the key zone, and the other samples will be loaded but remain unassigned.**

If the sample is already loaded but unassigned

1. **Select a key zone.**
 This can be empty, or contain a sample - it doesn't matter for now.
2. **Use the Sample knob to dial in the sample you want the key zone to play.**



The Sample knob.

Setting the Root Key

Once you have defined a key zone, and added a sample, you should set the root key for the sample.

- **Select the key zone the sample belongs to, and click on the key you wish to set the root key to.**
 Which key to select is normally determined by the pitch of the sample. For example if the sample plays a F#2 guitar note, click on F#2.
- **Note that it is possible to select a root key outside the key zone, if required.**

Removing sample(s) from a Key Map

- **To remove a sample, select the zone it belongs to, and then select "Delete Sample" from the Edit or context menus.**
 The sample is removed from the zone and from sample memory.
- **To remove a sample from a key zone/map, *without* removing it from memory, you can either select "No Sample" with the Sample knob for that zone, or simply replace it with another sample in the same way.**

Removing all unassigned samples

- **To remove all samples that are not assigned to any key zone, select Delete Unused Samples from the Edit menu.**

Rearranging samples in a Key Map

There is no specific function for rearranging or trading places between samples and key zones. Simply select a key zone and change the current sample assignment with the Sample knob.

Setting Sample Level

For each key zone you can set a volume level, using the Level button below the display. If the transition between two key zones causes a noticeable level difference, this parameter can be used to balance the levels.

Tuning samples

Sometimes you might find that the samples you wish to use in a key map are slightly out of tune with *each other*. This parameter allows you to tune each sample in a map by +/– half a semitone.

→ **Select the key zone(s) that contains the out of tune sample(s), and use the Tune knob below the keyboard display.**

– **If all samples originate from different sources, and all or most of them are pitched slightly different (a not uncommon sampling scenario), you could first tune them so that they all match each other, and then, if necessary, use the Sample Pitch controls in the Osc section to tune them globally to the "song" you wish to use the samples in.**

→ **Note that if all the samples were slightly out of tune *by the same amount* in relation to the song you intend to use the samples in, it would be much simpler to use the Sample Pitch controls in the Osc section directly.**

Looping Samples



A sample, unlike the cycles of an oscillator for example, is a finite quantity. There is a sample start and end. To get samples to play for as long as you press down the keys on your keyboard, they need to be *looped*.

For this to work properly, you have to first set up two loop points which determine the part of the sample that will be looped, and make this a part of the audio file. You cannot set loop points in the NN-19, this has to be done in an external sample editor.

All included samples already have set loop points (if needed).

For each sample (or key zone), you can select the following Loop modes by using the Loop knob below the keyboard display:

→ **OFF**
No looping is applied to the sample.

→ **FWD**
The part between the loop points plays from start to end, then the cycle is repeated. This is the most common loop mode.

→ **FWD - BW**
The part between the loop points plays from start to end, then from end to start, and then repeats the cycle.

! **For samples without any loop points, the whole sample will be looped.**

About the Solo Sample function

The Solo Sample button will allow you to listen to a selected sample over the entire keyboard range.

→ **Select the key zone the sample is assigned to, and then activate Solo Sample.**
This can be useful for checking if the root key is set correctly or if the current range is possible to extend etc.

! **For Solo Sample to work, "Select Key Zone via MIDI" must be disabled!**

Automap Samples

If you have a number of samples that belong together, but haven't mapped them to key zones you can use the "Automap Samples" function on the Edit menu. This is used in the following way:

1. **Select all samples that belong together and load them in one go, using the sample browser.**
 One of the samples will be assigned to a key zone spanning the whole range, and the rest will be loaded in to memory but remain unassigned.
2. **Select Automap Samples from the Edit menu.**

Now all samples currently in memory (assigned or unassigned) will be arranged automatically so that:

- **Each sample will be placed correctly according to its root note, and will be tuned according to the information in the sample file.**
 Most audio editing programs can save root key information as part of the file.
- **Each sample will occupy half the note range to the next sample's root note.**
 The root key will always be in the middle of each zone, with the zone extending both down and up in relation to the root position.

Mapping samples without Root Key or Tuning information

Some samples may not have any information about root key or tuning stored in the file. If the file names indicate the root key you can manually set it for each sample using the method described below. In a worst case scenario, i.e. no tuning or root key information whatsoever, you can still make use of the Automap function:

1. **Select all samples that belong together and load them in one go, using the sample browser.**
 One of the samples will be assigned to a key zone spanning the whole range, and the rest will be loaded in to memory but remain unassigned.
2. **Manually set the root key, and adjust the tune knob if the sample needs fine-tuning.**
 Without any information stored in the file, or if the file name doesn't indicate the root key, you will have to use your ears for this step. Play the sample and use another instrument or a tuner to determine its pitch.
3. **Select the next sample using the Sample knob, and repeat the previous step.**
 Proceed like this until you have set a root key for all the samples in memory.
4. **Select "Automap Samples" from the edit menu.**
 The samples will be mapped according to their set root key positions!

How Mapping Information is saved

All information about key zones, high and low range, root key etc. is stored as part of the Sampler Patch. The original sample files are never altered!

NN-19 synth parameters

The NN-19 synth parameters are used to shape and modulate samples. These are mostly similar to the parameters used to shape the oscillators in Subtractor - you have envelope generators, a filter, velocity control etc. Again, it is important to remember that these parameters do not alter the audio files in any way, only the way they will play back.

! **These parameters are global, in the sense that they will affect all samples in a sample patch.**

The Oscillator Section



For a sample patch, the actual samples are what oscillators are for a synthesizer, the main sound *source*. The following settings can be made in the Osc section of the NN-19:

Sample Start

This changes the start position of samples in a sample patch. Turning the knob clockwise gradually offsets the samples' start position, so that they will play back from a position further "into" the samples' waveform. This is useful mainly for two things:

- **Removing "air" or other unwanted artefacts from the start of less than perfect samples.**
 Occasionally (although not in any samples supplied with Reason) you may come across samples where the start point of the sample is slightly ahead of the start of the actual sound. There may be noise or silence in the beginning which was not intended to be part of the sample. By adjusting the sample start position, this can be removed.
- **Changing the start point as an effect.**
 For example, if you had a sample of someone saying "one, two, three", you could change the start position so that when you played the sample it would start on "three".

► **You can also assign velocity sample start allowing to use your playing to determine the exact sample start. See later in this chapter.**

Setting Sample Pitch - Octave/Semitone/Fine

By adjusting the corresponding knobs you can change the pitch of all samples belonging to a patch, in three ways:

- **Octave steps**
 The range is 0 - 8. The default setting is 4.
- **Semitone steps**
 Allows you to raise the frequency in 12 semitone steps (1 octave).

→ **Fine steps (100th of a semitone)**
The range is -50 to 50 (down or up half a semitone).

! **Note that the controls in this section cannot be used to tune samples against each other, as all samples will be affected equally. To tune individual samples, you use the Tune parameter below the keyboard display (see "Tuning samples").**

Keyboard Tracking

The Osc section has a button named "Kbd. Track". If this is switched off, the sample's pitch will remain constant, regardless of any incoming note pitch messages, although the oscillator still reacts to note on/off messages. This could be useful if you are using non-pitched samples, like drums for example. You could then play a sample in a zone using several keys, allowing for faster note triggering if you wanted to play a drum roll, for example.

Osc Envelope Amount

This parameter determines to what degree the overall pitch of the samples will be affected by the Filter Envelope (see "Filter Envelope"). You can set negative or positive values here, which determines whether an envelope parameter should raise or lower the pitch.

The Filter Section



Filters are used for shaping the overall timbre of the sound. The filter in NN-19 is a multimode filter with five filter types.

Filter Mode

With this selector you can set the filter to operate as one of five different types of filter. These are as follows:

- **24 dB Lowpass (LP 24)**
 Lowpass filters lets low frequencies pass and cuts out the high frequencies. This filter type has a fairly steep roll-off curve (24dB/Octave). Many classic synthesizers (Minimoog/Prophet 5 etc.) used this filter type.
- **12 dB Lowpass (LP 12)**
 This type of lowpass filter is also widely used in classic analog synthesizers (Oberheim, TB-303 etc.). It has a gentler slope (12 dB/Octave), leaving more of the harmonics in the filtered sound compared to the LP 24 filter.
- **Bandpass (BP 12)**
 A bandpass filter cuts both high and low frequencies, while midrange frequencies are not affected. Each slope in this filter type has a 12 dB/Octave roll-off.
- **High-Pass (HP12)**
 A highpass filter is the opposite of a lowpass filter, cutting out the lower frequencies and letting the high frequencies pass. The HP filter slope has a 12 dB/Octave roll-off.

- **Notch**
 A notch filter (or band reject filter) could be described as the opposite of a bandpass filter. It cuts off frequencies in a narrow midrange band, letting the frequencies below and above through.

Filter Frequency

The Filter Frequency parameter (often referred to as "cutoff") determines which area of the frequency spectrum the filter will operate in. For a lowpass filter, the frequency parameter could be described as governing the "opening" and "closing" of the filter. If the Filter Freq is set to zero, none or only the very lowest frequencies are heard, if set to maximum, all frequencies in the waveform are heard. Gradually changing the Filter Frequency produces the classic synthesizer filter "sweep" sound.

! **Note that the Filter Frequency parameter is usually controlled by the Filter Envelope (see "Envelope Section" below) as well. Changing the Filter Frequency with the Freq slider may therefore not produce the expected result.**

Resonance

The filter resonance parameter (sometimes called Q) is used to set the Filter characteristic, or quality. For lowpass filters, raising the filter Res value will emphasize the frequencies around the set filter frequency. This produces a generally thinner sound, but with a sharper, more pronounced filter frequency "sweep". The higher the resonance value, the more resonant the sound becomes until it produces a whistling or ringing sound. If you set a high value for the Res parameter and then vary the filter frequency, this will produce a very distinct sweep, with the ringing sound being very evident at certain frequencies.

- **For the highpass filter, the Res parameter operates just like for the lowpass filters.**
- **When you use the Bandpass or Notch filter, the Resonance setting adjusts the width of the band. When you raise the Resonance, the band where frequencies are let through (Bandpass), or cut (Notch) will become narrower. Generally, the Notch filter produces more musical results using low resonance settings.**

Envelope Section

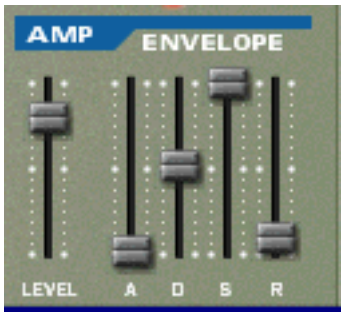
Envelope generators are used to control several important sound parameters in analog synthesizers, such as pitch, volume, filter frequency etc. Envelopes govern how these parameters should respond over time - from the moment a note is struck to the moment it is released.

Standard synthesizer envelope generators have four parameters; Attack, Decay, Sustain and Release (ADSR).

There are two envelope generators in the NN-19, one for volume, and one for the filter frequency.

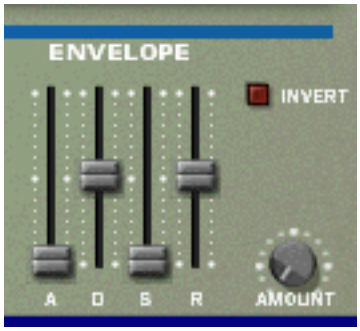
! **Please refer to the Subtractor chapter for a description of the basic envelope parameters.**

Amplitude Envelope



The Amp Envelope is used to adjust how the volume of the sound should change from the time you press a key until the key is released. By setting up a volume envelope you sculpt the sound's basic shape with the four Amplitude Envelope parameters, Attack, Decay, Sustain and Release. This determines the basic character of the sound (soft, long, short etc.). The Level parameter acts as a general volume control for the sample patch.

Filter Envelope



The Filter Envelope can be used to control two parameters; filter frequency and sample pitch. By setting up a filter envelope you control the how the filter frequency and/or the sample pitch should change over time with the four Filter Envelope parameters, Attack, Decay, Sustain and Release.

Filter Envelope Amount

This parameter determines to what degree the filter will be affected by the Filter Envelope. Raising this knob's value creates more drastic results. The Envelope Amount parameter and the set filter frequency are related. If the Filter Freq slider is set to around the middle, this means that the moment you press a key the filter is already halfway open. The set Filter Envelope will then open the filter further from this point. The Filter Envelope Amount setting affects how much further the filter will open.

Filter Envelope Invert

If this button is activated, the envelope will be inverted. For example, normally the Decay parameter lowers the filter frequency, but after activating Invert it will instead raise it, by the same amount. Note that Invert does not affect the Osc pitch parameter (this can be inverted by setting positive or negative values).

LFO Section



LFO stands for Low Frequency Oscillator. LFOs are oscillators in the sense that they generate a waveform and a frequency. However, there are two significant differences compared to normal sound generating oscillators:

- **LFOs only generate waveforms with low frequencies.**

- **The output of the two LFOs are never actually heard. Instead they are used for modulating various parameters.**

The most typical application of an LFO is to modulate the pitch of a (sound generating) oscillator or sample, to produce vibrato.

The LFO section has the following parameters:

Waveform

LFO 1 allows you to select different waveforms for modulating parameters. These are (from the top down):



Waveform	Description
Triangle	This is a smooth waveform, suitable for normal vibrato.
Inverted Sawtooth	This produces a “ramp up” cycle. If applied to an oscillator’s frequency, the pitch would sweep up to a set point (governed by the Amount setting), after which the cycle immediately starts over.
Sawtooth	This produces a “ramp down” cycle, the same as above but inverted.
Square	This produces cycles that abruptly changes between two values, usable for trills etc.
Random	Produces random stepped modulation to the destination. Some vintage analog synths called this feature “sample & hold”.
Soft Random	The same as above, but with smooth modulation.

Destination

The available LFO Destinations are as follows:

Destination	Description
Osc	Selecting this makes LFO control the pitch (frequency) of the sample patch.
Filter	Selecting this makes the LFO control the filter frequency.
Pan	Selecting this makes the LFO modulate the pan position of samples, i.e. it will move the sound from left to right in the stereo field.

Sync

By clicking this button you activate/deactivate LFO sync. The frequency of the LFO will then be synchronized to the song tempo, in one of 16 possible time divisions. When sync is activated, the Rate knob (see below) is used for setting the desired time division.

Turn the knob and check the tooltip for an indication of the time division.



Rate

The Rate knob controls the LFO's frequency. Turn clockwise for a faster modulation rate.

Amount

This parameter determines to what degree the selected parameter destination will be affected by the LFO. Raising this knob's value creates more drastic results.

Play Parameters

This section deals with two things: Parameters that are affected by how you play, and modulation that can be applied manually with standard MIDI keyboard controls.

These are:

- **Velocity Control**
- **Pitch Bend and Modulation Wheel**
- **Legato**
- **Portamento**
- **Polyphony**
- **Voice Spread**
- **External Controllers**

Velocity Control



Velocity is used to control various parameters according to how hard or soft you play notes on your keyboard. A common application of velocity is to make sounds brighter and louder if you strike the key harder. By using the knobs in this section, you can control how much the various parameters will be affected by velocity. The velocity sensitivity amount can be set to either positive or negative values, with the center position representing no velocity control.

The following parameters can be velocity controlled:

Destination	Description
Amp	This let's you velocity control the overall volume of the sound. If a positive value is set, the volume will increase the harder you strike a key. A negative value inverts this relationship, so that the volume decreases if you play harder, and increases if you play softer. If set to zero, the sound will play at a constant volume, regardless of how hard or soft you play.
F. Env	This sets velocity control for the Filter Envelope Amount parameter. A positive value will increase the envelope amount the harder you play. Negative values invert this relationship.
F. Dec	This sets velocity control for the Filter Envelope Decay parameter. A positive value will increase the Decay time the harder you play. Negative values invert this relationship.
S.Start	This sets velocity control for the Sample Start parameter. A positive value will increase the Start Time amount the harder you play. Negative values invert this relationship.
A. Attack	This sets velocity control for the Amp Envelope Attack parameter. A positive value will increase the Attack time the harder you play. Negative values invert this relationship.

Pitch Bend and Modulation Wheels

The Pitch Bend wheel is used for "bending" notes, like bending the strings on a guitar. The Modulation wheel can be used to apply various modulation while you are playing. Virtually all MIDI keyboards have Pitch Bend and Modulation controls. NN-19 also has two functional wheels that could be used to apply real time modulation and pitch bend should you not have these controllers on your keyboard, or if you aren't using a keyboard at all. The wheels mirror the movements of the MIDI keyboard controllers.

Pitch Bend Range

The Range parameter sets the amount of pitch bend when the wheel is turned fully up or down. The maximum range is "24" (=up/down 2 Octaves).

Modulation Wheel

The Modulation wheel can be set to simultaneously control a number of parameters. You can set positive or negative values, just like in the Velocity Control section. The following parameters can be affected by the modulation wheel:

Destination	Description
F. Freq	This sets modulation wheel control of the Filter Frequency parameter. A positive value will increase the frequency if the wheel is pushed forward. Negative values invert this relationship.
F. Res	This sets modulation wheel control of the Filter Resonance parameter. A positive value will increase the resonance if the wheel is pushed forward. Negative values invert this relationship.
F. Dec	This sets modulation wheel control for the Filter Envelope Decay parameter. A positive value will increase the decay if the wheel is pushed forward. Negative values invert this relationship.
LFO	This sets modulation wheel control of the LFO Amount parameter. A positive value will increase the Amount if the wheel is pushed forward. Negative values invert this relationship.
Amp	This sets modulation wheel control for the Amp level parameter. A positive value will increase the level if the wheel is pushed forward. Negative values invert this relationship.

Legato

Legato works best with monophonic sounds. Set Polyphony (see "Setting Number of Voices - Polyphony") to 1 and try the following:

- **Hold down a key and then press another key without releasing the previous.**
 Notice that the pitch changes, but the envelopes do not start over. That is, there will be no new "attack".
- **If polyphony is set to more voices than 1, Legato will only be applied when all the assigned voices are "used up".**
 For example, if you had a polyphony setting of "4" and you held down a 4 note chord, the next note you played would be Legato. Note, however, that this Legato voice will "steal" one of the voices in the 4 note chord, as all the assigned voices were already used up!

Retrig

This is the "normal" setting for playing polyphonic patches. That is, when you press a key without releasing the previous, the envelopes are retriggered, like when you release all keys and then press a new one. In monophonic mode, Retrig has an additional function; if you press a key, hold it, press a new key and then release that, the first note is also retriggered.

Portamento (Time)

Portamento is when the pitch "glides" between the notes you play, instead of instantly changing the pitch. The Portamento knob is used to set how long it takes for the pitch to glide from one pitch to the next. If you don't want any Portamento at all, set this knob to zero.

Setting Number of Voices - Polyphony

This determines the polyphony, i.e. the number of voices a patch can play simultaneously. This can be used to make a patch monophonic (=a setting of "1"), or to extend the number of voices available for a patch. The maximum number of voices you can set a patch to use is 99.

! **Note that the Polyphony setting does not "hog" voices. For example, if you have a patch that has a polyphony setting of ten voices, but the part the patch plays only uses four voices, this won't mean that you are "wasting" six voices. In other words, the polyphony setting is not something you need to consider if you want to conserve CPU power - it is only the number of voices *actually used* that counts.**

Voice Spread

This parameter can be used to control the stereo (pan) position of voices. The Spread knob determines the intensity of the panning. If this is set to "0", no panning will take place. The following pan modes can be selected:

Mode	Description
Key	This will shift the pan position gradually from left to right the higher up on the keyboard you play.
Key 2	This will shift the pan position from left to right in 8 steps (1/2 octave) for each consecutive higher note you play, and then repeat the cycle.
Jump	This will alternate the pan position from left to right for each note played.

Low Bandwidth

This will remove some high frequency content from the sound, but often this is not noticeable (this is especially true if you have "filtered down" samples). Activating this mode will save you some extra computer power, if needed.

Controller Section

NN-19 can receive common MIDI controller messages, and route these to various parameters. The following MIDI messages can be received:

- **Aftertouch (Channel Pressure)**
- **Expression Pedal**
- **Breath Control**

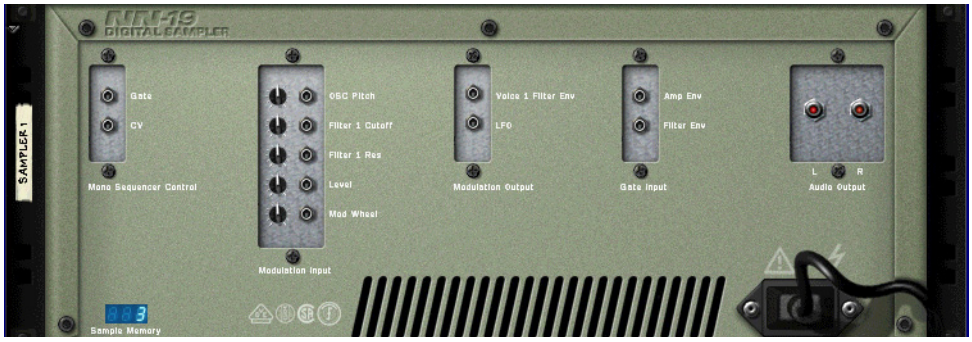
If your MIDI keyboard is capable of sending Aftertouch messages, or if you have access to an Expression Pedal or a Breath controller, you can use these to modulate NN-19 parameters. The "Source" selector switch determines which of these message-types should be received.

These messages can then be assigned to control the following parameters:

F. Freq	This sets external modulation control of the filter frequency parameter. A positive value will increase the frequency with higher external modulation values. Negative values invert this relationship.
LFO 1	This sets external modulation control of the LFO Amount parameter. A positive value will increase the LFO amount with higher external modulation values. Negative values invert this relationship.
Amp	This let's you control the overall volume of the sound with external modulation. If a positive value is set, the volume will increase with higher external modulation values. A negative value inverts this relationship.

Connections

On the back panel of the NN-19 you will find the connectors, which are mostly CV/Gate related.



Audio Outputs

These are the main left and right audio outputs. When you create a new NN-19 device, these are auto-routed to the first available outputs in the I/O device.

Mono Sequencer Control

These are the main CV/Gate inputs. CV controls the note pitch. Gate inputs trigger note on/off values plus a level, which can be likened to a velocity value. If you want to control the NN-19 from a Matrix Pattern Sequencer for example, you would normally use these inputs. The inputs are “mono”, i.e. they control one voice in the sampler.

Modulation Inputs

! **Remember that CV connections will not be stored in the sample patch, even if the connections are to/from the same NN-19 device!**

These control voltage (CV) inputs (with associated voltage trim pots), can modulate various NN-19 parameters from other devices, or from the modulation outputs of the same NN-19 device. These inputs can control the following parameters:

- **Osc (sample) Pitch**
- **Filter Cutoff**
- **Filter Resonance**
- **Amp Level**
- **Mod Wheel**

Modulation Outputs

The Modulation outputs can be used to voltage control other devices, or other parameters in the same NN-19 device. The Modulation Outputs are:

- **Filter Envelope**
- **LFO**

Gate Inputs

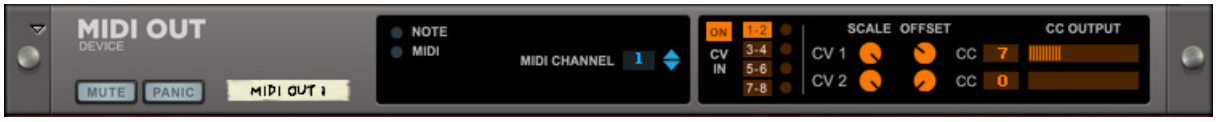
These inputs can receive a CV signal to trigger the envelopes. Note that connecting to these inputs will override the "normal" triggering of the envelopes. For example, if you connected a LFO output to the Gate Amp input, you would not trigger the amp envelope by playing notes, as this is now controlled by the LFO. In addition you would only hear the LFO triggering the envelope for the notes that you hold down.

- **Amp Envelope**
- **Filter Envelope**

Chapter 28
MIDI Out Device



Introduction



The MIDI Out Device is designed for routing MIDI out of the Reason Rack Plugin instance to other tracks/destinations in your main DAW. A typical scenario would be to route MIDI from a Player device in Reason Rack Plugin to another instrument plugin in your song/project.

The MIDI Out Device does not produce any sound of its own; it only directs MIDI from the Reason Rack Plugin instance to a selected MIDI Channel.

Using the MIDI Out Device

Setting up for MIDI controlling an external track/plugin

In this example we will route the MIDI from a Dual Arpeggio Player device via a MIDI Out Device.

1. **Drag a MIDI Out Device from the Instruments palette in the Browser and drop in the rack.**
 The MIDI Out Device is created in the rack.
2. **Click the Players palette and drag a Dual Arpeggio Player and drop above the MIDI Out Device:**



 The Dual Arpeggio Player is automatically attached to the MIDI Out Device.
3. **Create a MIDI track in the DAW sequencer.**
4. **Select Reason Rack Plugin as MIDI Input port for that track (refer to the DAW manual).**

! **With most DAWs, a Reason Rack Plugin instance can produce both audio and MIDI at the same time (i.e. a Reason Rack Plugin instrument can also send out MIDI). However, with DAWs using the AU plugin format (e.g. Logic), you need to add Reason Rack Plugin in a special MIDI FX slot for it to output MIDI. That instance will be a MIDI effect only and won't output audio.**

! **If you get problems with "hanging" notes, click the Panic button to send out an "All Notes Off".**



► **To route MIDI from other Player devices in the Reason Rack Plugin instance, simply create another MIDI Out Device and attach another Player to it. Then, select a different MIDI Channel on the MIDI Out Device.**

- **You don't have to attach a Player to the MIDI Out Device - you could use the MIDI Out Device just for "through-put" of the MIDI from the Reason Rack Plugin instance.**
 In these situations it doesn't matter where in the rack you place the MIDI Out Device.

About recording the Player MIDI from the MIDI Out Device

If you want to record the MIDI from the Player, you will have to do that in real-time on the destination track in your host DAW, see "Getting the Player MIDI output onto a track in your DAW".

Modulating MIDI Controllers from CV signals

The MIDI Out Device features eight CV inputs for routing modulation signals from the Reason Rack Plugin instance to MIDI CC# of your choice. These MIDI CC# changes are then transmitted on the selected MIDI Channel of the MIDI Out Device.

1. **Flip the rack around and connect some modulation sources to the desired CV inputs:**



2. **Flip the rack back to the front and click the On button to activate the CV IN section:**



 CV signals routed to any of the four CV IN pairs are indicated by lit LEDs:



3. **Click to select which CV IN pair to edit:**



4. **Drag up/down to select the desired MIDI CC# to route the CV modulation signals to:**



5. **Turn the Scale knobs to change the modulation range, from static (0-0) to full (0-127):**



6. **Turn the Offset knobs to change the modulation offset (0-127):**



The modulation level(s) are shown in the CC Output displays:



7. **Repeat steps 3-6 to assign and set up the other CV IN pairs.**

- **CV can be bipolar (have negative or positive values) but MIDI CC values are always positive. Any negative CV values will be truncated to zero when converted to MIDI CC. To preserve the shape of e.g. a modulation LFO, you can use the Scale and Offset controls to convert the bipolar CV (going between -127 and +127) to a MIDI CC going between 0 and 127. In that case, set both Scale and Offset halfway up.**
- **You can also use the Offset knobs for manually setting MIDI CC values without anything connected to the CV inputs.**
- **The CV Input ON button enables CV inputs. Turn this off if you want to make manual settings, select MIDI CCs etc, without the values being modulated by CV.**

Connections



Sequencer Control

The Sequencer Control CV In and Gate In inputs allow you to play the MIDI Out Device from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV In controls the note pitch, while the signal to the Gate In delivers note on/off along with velocity. There are also inputs for modulating the Pitch Bend and ModWheel parameters.

CV In to MIDI CC Out

These eight CV inputs can be used for modulating the desired MIDI CC#. The affected MIDI CC#s are defined on the front panel, see "Modulating MIDI Controllers from CV signals".

Chapter 29
Quartet
Chorus Ensemble

Introduction



Quartet Chorus Ensemble is a fabulous sounding chorus device, with four different characteristic chorus/ensemble algorithms. Each of the four algorithms can have their own unique parameter settings - including the Dry/Wet parameter - so you could switch between the algorithms and get the exact result you are looking for.

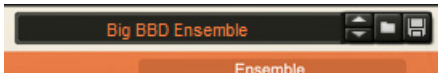
Quartet is designed to be used mainly as an insert effect, for spicing up individual instrument sounds with nice dense choruses and modulations.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Panel reference

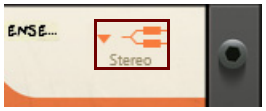
Global controls

Loading and saving patches



Loading and saving patches is done in the same way as with any other internal Reason device, see "Loading patches" and "Saving patches" for details.

Routing



→ **Click the Routing selector to select "Stereo" or Dual Mono" from the pop-up menu.**
Stereo: With this selected the L+R input signals are mixed before being sent into the stereo effect. This means you can connect a mono input signal and get stereo output signals.

Dual Mono: The L+R input channels are processed independently.

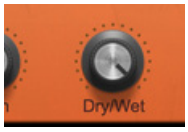
Width



The Width control can be used for setting the stereo width - from mono to nice and wide stereo.

! **Note that the Width parameter can be set individually for each of the four chorus algorithms.**

Dry/Wet



This controls the mix of the dry and processed signals.

! **Note that the Dry/Wet parameter can be set individually for each of the four chorus algorithms.**

! **Note that all chorus/ensemble effects require some amount of dry signal to produce the desired effect. Therefore, 100% Wet also includes a certain amount of dry signal.**

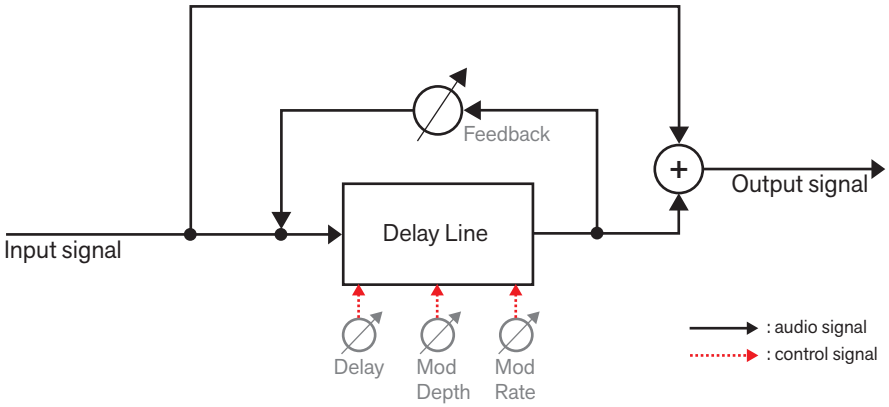
► **If you are using Quartet as a send effect you would probably want to have the Dry/Wet knob set to 100%.**

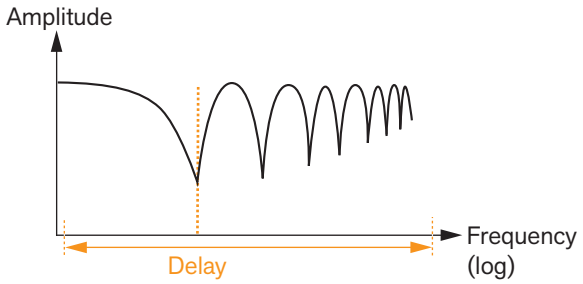
Chorus



The Chorus effect algorithm simulates multiple detuned "copies" of the input signal. The Chorus is basically a delay line with adjustable feedback. The principle is to split the input signal in two, run one signal dry and the other through the delay line, and then sum the two signals.

The picture below shows the basic principle of the chorus:





Delay



Here you set the delay time between the dry and processed signals. In practice, this determines where the notches/peaks will appear in the frequency spectrum.

Range: 1.00-30.00 ms

Mod Depth



This determines the depth of the LFO modulation, i.e. by how much the delay time should be modulated. If you set this to 0, the delay time will be static (most effective if you add some feedback).

Mod Rate



This determines the frequency of the LFO modulating the delay time. The higher the value, the faster the sound will oscillate.

Range: 0.10-5.00 Hz

Feedback



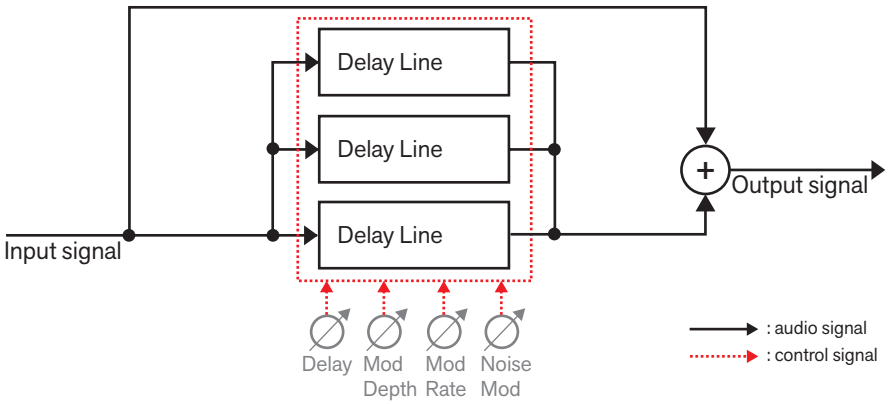
This governs the amount of effect signal fed back to the input, which in turn affects the intensity and character of the chorus effect. Turning this towards 100% produces a flanger type of effect with a pronounced resonance “tone”, while keeping it around 50% produces a more gentle chorus effect.

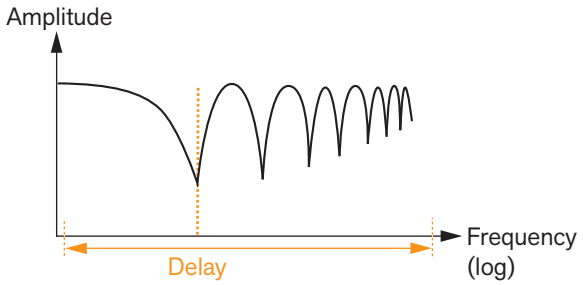
BBD



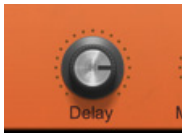
The BBD is a bucket brigade delay line which simulates vintage ensemble effects. Historically, the bucket brigade delay line was built up by a series of (analog) capacitors, that were clocked to consecutively transmit signals, via one capacitor at a time, thus creating a delayed signal. The BBD algorithm in Quartet features three chorus effects in parallel, and therefore provides a much richer and denser effect than the Chorus algorithm.

The picture below shows the basic principle of the BBD algorithm:





Delay



Here you set the delay time between the dry and processed signals. The delay is preset scaled between the three delay lines. In practice, this determines where the notches/peaks will appear in the frequency spectrum.

Range: 1.00-30.00 ms

Mod Depth



This determines the depth of the LFO modulation, i.e. by how much the delay time should be modulated. If you set this to 0, the delay time will be static (unless you are using Noise Mod, see "Noise Mod").

! **If Mod Depth and Noise Mod (see "Noise Mod") are both set to 0, the Width control (see "Width") has no effect.**

Mod Rate



This determines the frequency of the LFO modulating the delay time.

Range: 0.20-10.00 Hz

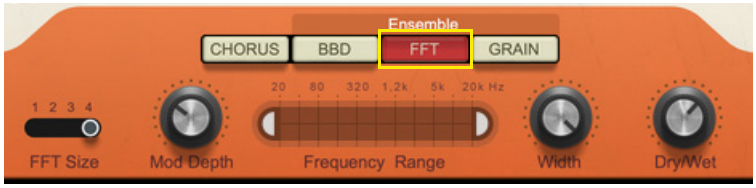
Noise Mod



This amplitude-modulates the signal with lowpass filtered noise, and generates a kind of "sparkling" effect.

! **If Noise Mod and Mod Depth (see "Mod Depth") are both set to 0, the Width control (see "Width") has no effect.**

FFT



The FFT algorithm simulates a type of chorus/ensemble effect by utilizing noise modulation of the signal partials. First the signal is analyzed using FFT (Fast Fourier Transform) and converted to a representation in the frequency domain. Then, the partials are modulated by noise to achieve a very nice and dense ensemble effect.

FFT Size



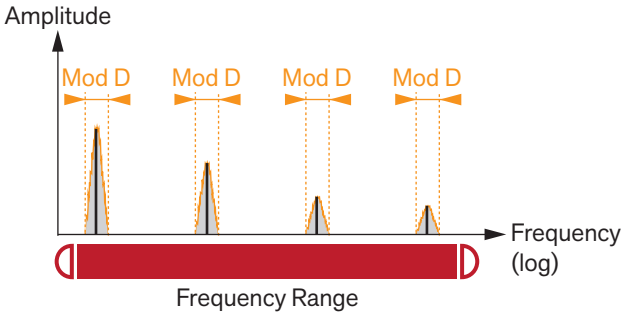
This sets the accuracy (and speed) of the frequency analysis. "1" is the fastest detection and preserves transients in the signal - but this also leaves out detection of low frequencies. "4" is the most accurate detection. However, it's also slower since it also detects low-frequency material (which takes a little longer to detect).

Mod Depth



This determines the depth of the noise modulation of the signal's partials. The parameter controls a combination of noise amplitude and bandwidth. The result also depends on the Frequency Range parameter (see "Frequency Range").

The picture below shows how the Mod Depth parameter affects the partials at full Frequency Range:

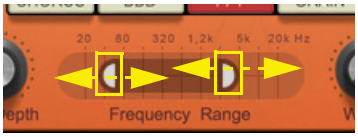


Frequency Range

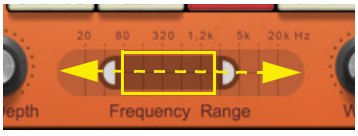


The Frequency Range parameter determines which part of the frequency range should be noise-modulated and which part should be left unaffected.

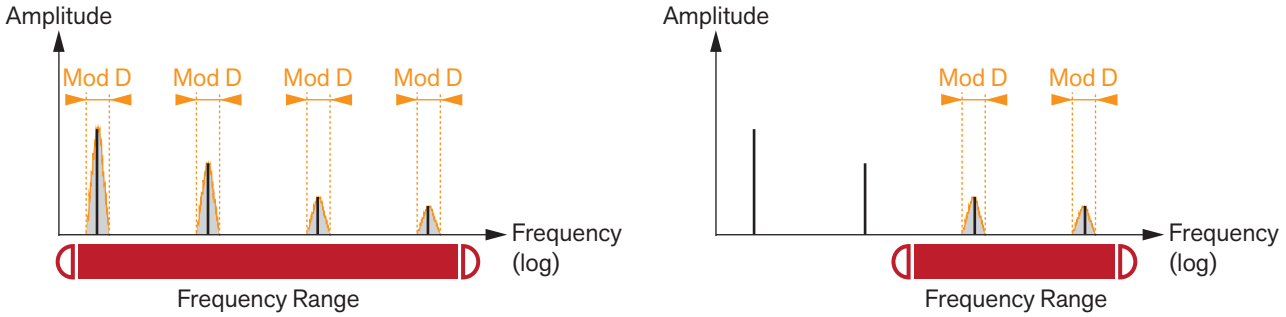
→ **Set the desired Frequency Range by dragging either "handle" sideways.**



→ **To move the Frequency Range while maintaining the currently set bandwidth, drag the area between the "handles" sideways.**

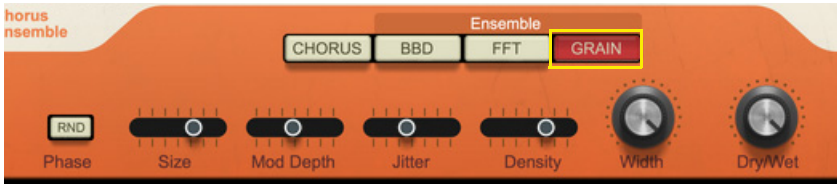


The picture below shows how the Mod Depth parameter affects the partials in the signal at two different Frequency Range settings:

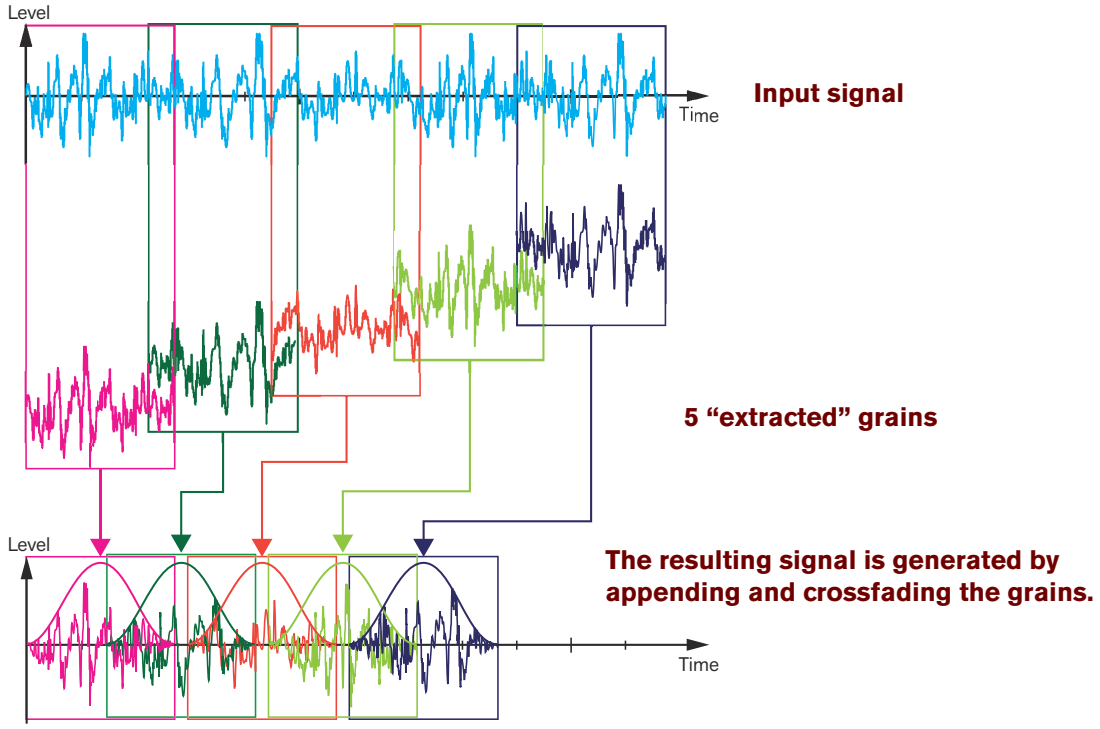


The first example shows the modulation of the partials at full bandwidth. The second example shows the partial modulation with the lower Frequency set to a higher value. In the second example, only the upper partials are modulated. The lower partials are left unaffected.

Grain



The Grain algorithm generates an ensemble effect by "extracting" grains from the input signal in real-time and then cross-fading through the grains in various ways. The method is similar to the "Long Grains" algorithm used in the Grain Sample Manipulator device in Reason. The picture below shows the principle for the Grain algorithm:



An example of a signal generated from 5 grains of the input signal.

Phase



The Random Phase function randomly alters the phase of the grains to create a “bubbly” kind of effect, caused by phase cancellation. This is especially noticeable when the Jitter parameter (see “Jitter”) is set to a low value.

Size



This controls the grain length. High values produce a more smooth effect, whereas low values generate more of a “stuttering” effect.

Mod Depth



This randomly changes the initial pitch of the grains.

Jitter



The Jitter function modulates the grain playback position randomly. The Jitter function can be great for generating chorus-like effects and to make a sound more “alive”, depending on the other settings.

Density



The Density function is a combination of grain size, playback rate and the amount of grain overlap. High values produce a really fat and dense chorus/ensemble effect, whereas low values generate a “thinner” effect.

Connections



CV Input

Mod Depth

This CV input can be used for modulating the Mod Depth parameter in the different algorithms. The input accepts bipolar control signals. The input signal can be attenuated with the corresponding attenuation knob.

! **Note that the CV Modulation is global for the Mod Depth control in all algorithms.**

Width

This CV input can be used for modulating the Width parameter in the different algorithms. The input accepts bipolar control signals. The input signal can be attenuated with the corresponding attenuation knob.

! **Note that the CV Modulation is global for the Width control in all algorithms.**

Dry/Wet

This CV input can be used for modulating the Dry/Wet parameter in the different algorithms. The input accepts bipolar control signals. The input signal can be attenuated with the corresponding attenuation knob.

! **Note that the CV Modulation is global for the Dry/Wet control in all algorithms.**

Input Left & Right

→ **Patch the audio signals you want to process here.**
If your input signal is in mono, connect only to the L (left) input.

! **Note that the result also depends on the current Routing setting (see "Routing").**

Output Left & Right

These are the audio outputs.

! **Note that the result also depends on the current Routing setting (see "Routing").**

Chapter 30
Sweeper
Modulation Effect

Introduction



The Sweeper Modulation Effect device is a great sounding Phaser/Flanger/Filter device.

By drawing your own unique modulation curve in the display and assigning this curve to the desired effect parameters, you also get a very flexible system for repeatedly sweeping/modulating the effects parameters - in perfect sync with the sequencer.

Sweeper is designed to be used mainly as an insert effect, for spicing up instrument sounds with nice sweeps and modulations, but you could of course use it as you like!

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Panel reference

Global controls

Loading and saving patches



Loading and saving patches is done in the same way as with any other internal Reason device, see “Loading patches” and “Saving patches” for details.

Volume



This is the master volume control.

Routing



→ **Click the Routing selector to select "Stereo" or Dual Mono" from the pop-up menu.**
Stereo: With this selected the L+R input signals are mixed before being sent into the stereo effect. This means you can connect a mono input signal and get stereo output signals.

Dual Mono: The L+R input channels are processed independently.

Spread



The Spread control detunes the stereo channels to generate a nice and wide stereo effect. Note, though, that the Spread control works a little differently and has different ranges in the Phaser, Flanger and Filter.

Dry/Wet



This controls the mix of the dry and processed signals.

LFO -> Freq



This controls the modulation amount from the LFO (see "LFO") to the Frequency control of the Phaser (see "Frequency"), Flanger (see "Frequency") and Filter (see "Frequency") section.

The control is bipolar, which means that negative values will invert the modulation.

Mod -> Freq



This controls the modulation amount from the Modulator (see "The Envelope Modulator" and "The Audio Follower Modulator") to the Frequency control of the Phaser (see "Frequency"), Flanger (see "Frequency") and Filter (see "Frequency") section.

The control is bipolar, which means that negative values will invert the modulation.

LFO -> Volume



This controls the modulation amount from the LFO (see "LFO") to a separate built-in amplifier. The control is bipolar, which means that negative values will invert the modulation.

Mod -> Volume



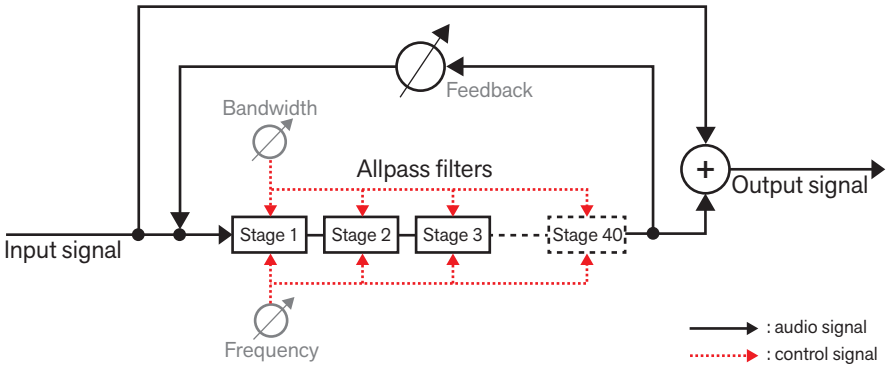
This controls the modulation amount from the Modulator (see "The Envelope Modulator" and "The Audio Follower Modulator") to a separate built-in amplifier.

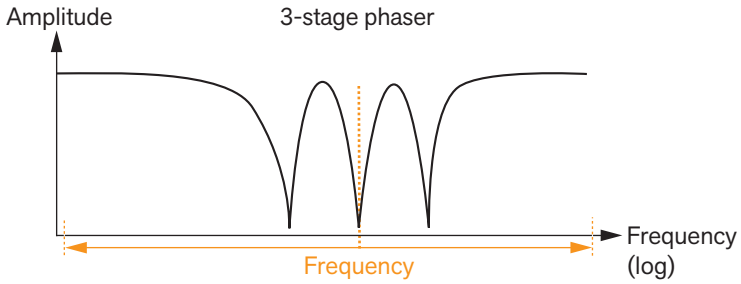
The control is bipolar, which means that negative values will invert the modulation.

The Phaser



The Phaser consists of a number of all-pass filters (1 to 40) with feedback, which can be used for creating really nice phasing effects. An all-pass filter lets all frequencies of a signal through - but phase inverted 180 degrees. The principle is to split the input signal in two, run one signal dry and the other through a series of all-pass filters, and then sum the two signals. The picture below shows the basic principle of a phaser:





Frequency



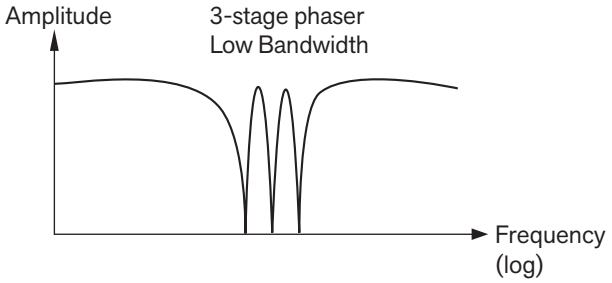
Here you set the frequency of the all-pass filter(s) in the phaser.

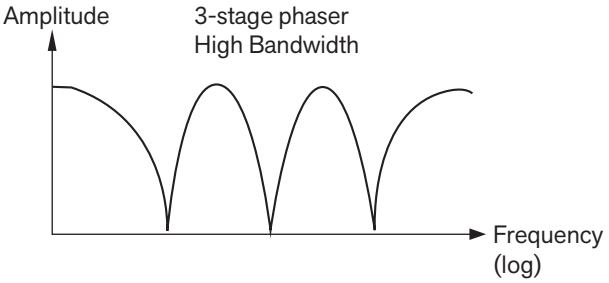
Range: 37.6 Hz to 16.17 kHz

Bandwidth



This controls the bandwidth of the all-pass filter(s) in the phaser.

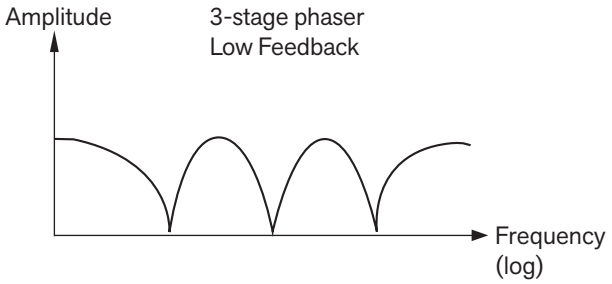


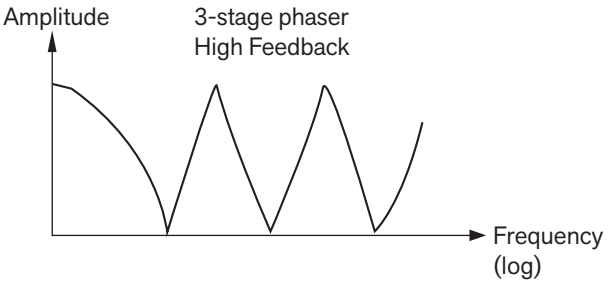


Feedback



This controls the level/intensity of the phaser peaks and notches.



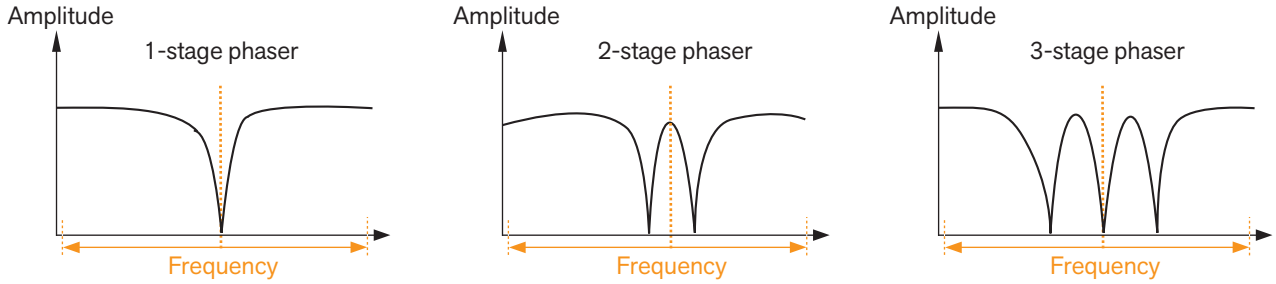


Stages



In a phaser, a stage (also known as “pole”) is represented by an all-pass filter. Here you set the number of all-pass filters you want to use. Each all-pass filter contributes with one notch/peak in the frequency spectrum.

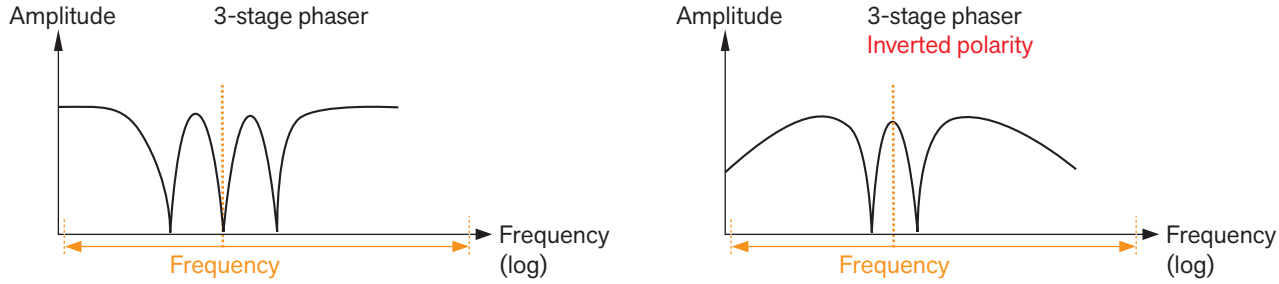
Range: 1-40 stages (notches)



Polarity



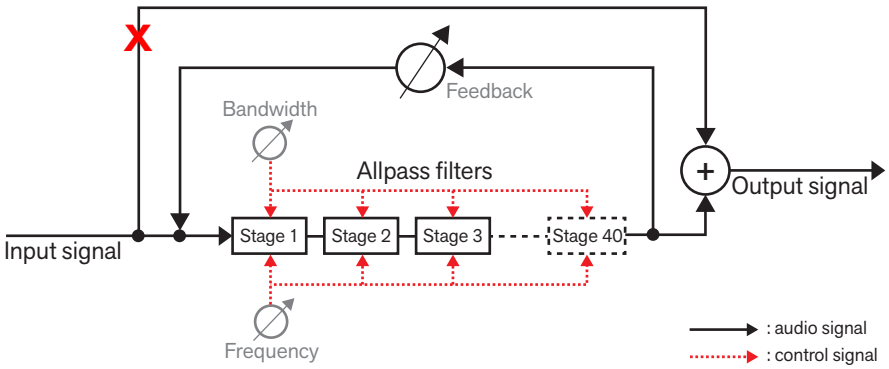
Pressing this button will invert the polarity of the Phaser filter, so that instead of notches in the frequency spectrum, there will be peaks:



Mute Dry



Pressing this button mutes the dry signal in the Phaser section, turning the effect into a frequency-dependent delay. Since no dry signal is mixed with the effect signal, there will be no notches in the frequency spectrum:

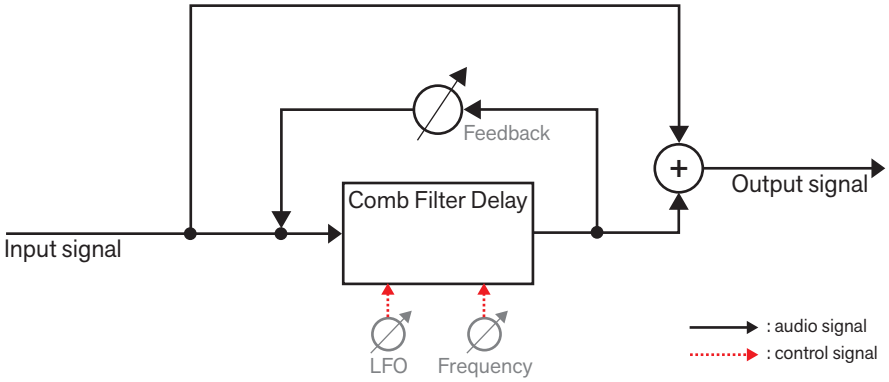


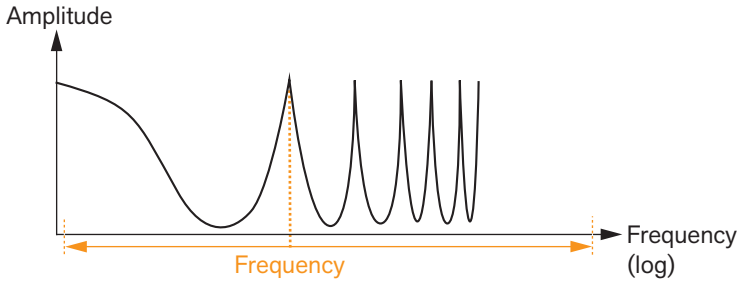
This will give more of a “tremolo” effect rather than phasing.

The Flanger



The Flanger is basically a Comb Filter with adjustable feedback, which can be used for creating a wide variety of chorus effects and frequency swirls. The principle is to split the input signal in two, run one signal dry and the other through a comb filter delay, and then sum the two signals. The picture below shows the basic principle of a flanger:





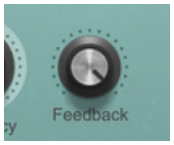
Frequency



Here you set the comb filter frequency (in practice, the delay time between the dry and processed signals).

Range: 37.6 Hz to 16.17 kHz

Feedback

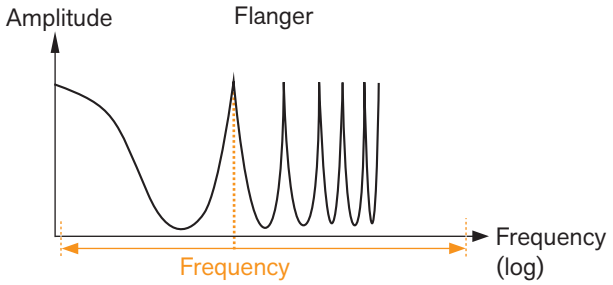


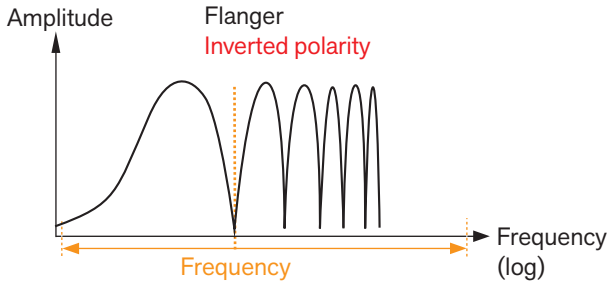
This intensifies the flange effect by increasing the resonance peaks via feedback.

Polarity

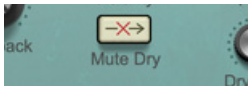


Pressing this button will invert the polarity of the Flanger, so that instead of peaks in the frequency spectrum, there will be notches:

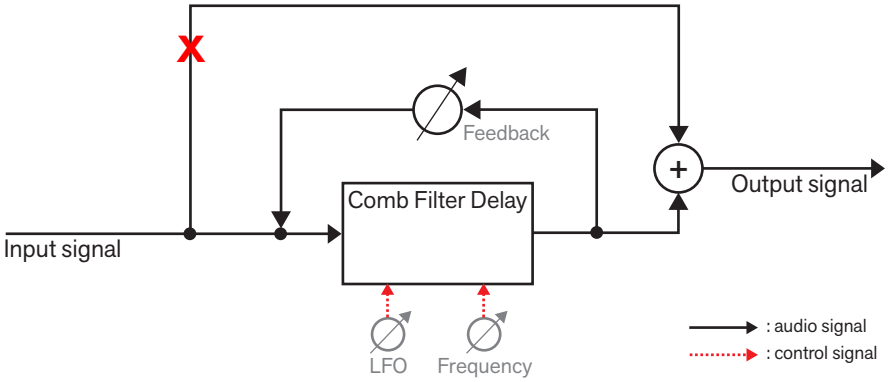




Mute Dry



Pressing this button mutes most of the dry signal in the Flanger section:



The Filter



The Filter section features a selection of great sounding filters with various characteristics, derived from the Europa Shapeshifting Synthesizer.

Drive



This amplifies and introduces an overdrive type of distortion to the signal in the filter.

Frequency



Here you set the cutoff frequency (for the HP and LP filter types) or the center frequency (for the BP and Notch filter types).

Resonance



This controls the resonance amount, i.e. the amplification of the signal around the cutoff frequency.

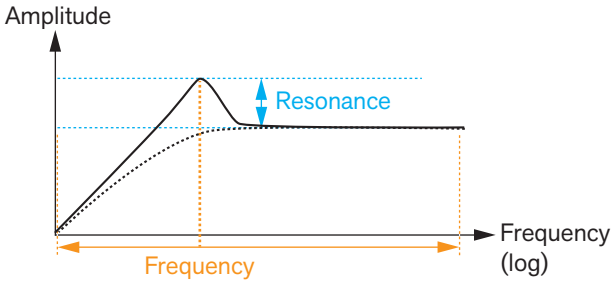
! **In the SVF Notch filter, the Resonance knob controls the width of the notch - from wide to narrow.**

(Filter) Type



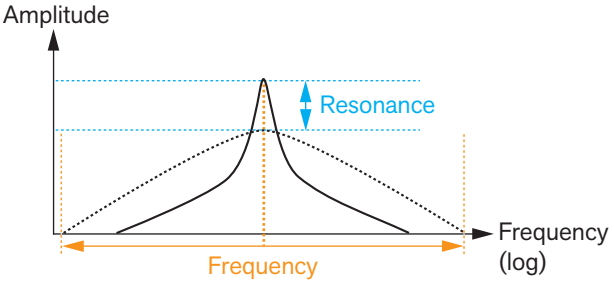
→ **Click the TYPE selector to select one of the following filter types from the pop-up menu:**

- **SVF HP 12dB**



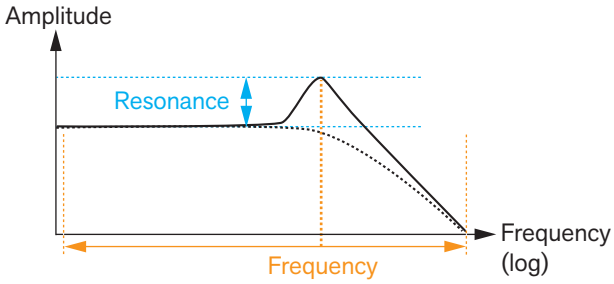
A state variable (SVF) highpass filter with a 12dB/octave slope.

- **SVF BP 12dB**



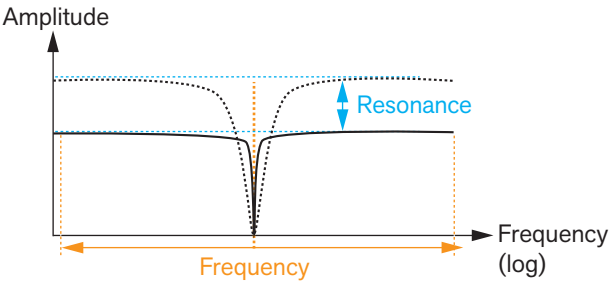
A state variable (SVF) bandpass filter with 12dB/octave slopes.

- **SVF LP 12dB**



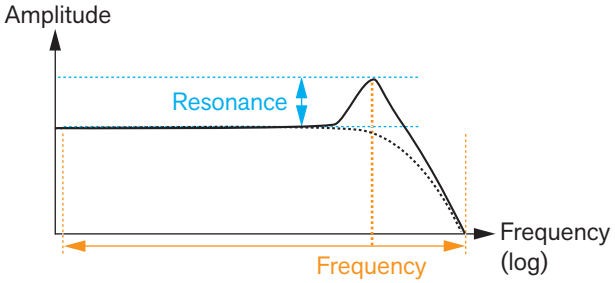
A state variable (SVF) lowpass filter with a 12dB/octave slope.

- **SVF Notch**



A state variable (SVF) notch filter.

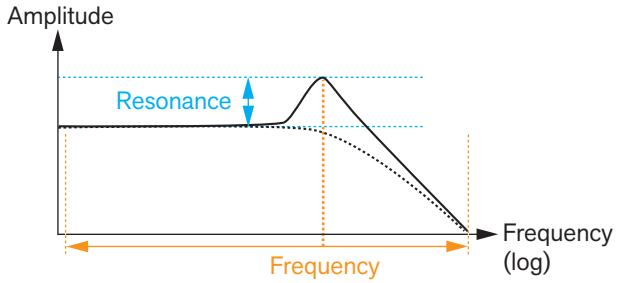
- **Ladder LP 24dB**



A ladder-type lowpass filter with a 24dB/octave slope. The resonance peak more narrow in this filter type than in the MFB LP 24dB filter (see below). The filter can be driven to self-oscillate.

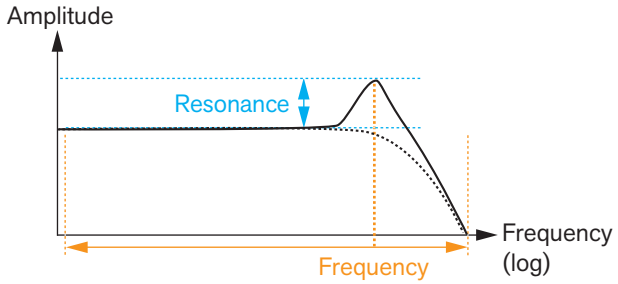
! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

- **MFB LP 12dB**



A multiple feedback (MFB) lowpass filter with a 12dB/octave slope. If you turn up the Resonance high, additional resonance peaks appear.

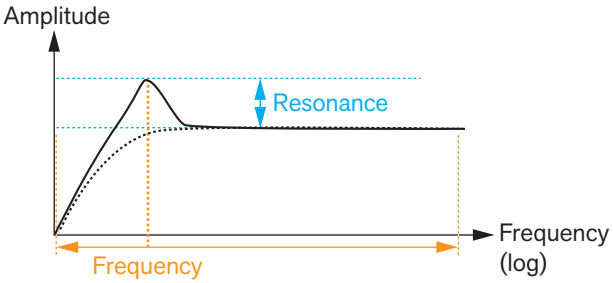
- **MFB LP 24dB**



A multiple feedback (MFB) lowpass filter with a 24dB/octave slope. The resonance peak is wider in this filter type that in the Ladder filter (see above). The filter can be driven to self-oscillate. If you turn up the Resonance high, additional resonance peaks appear.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

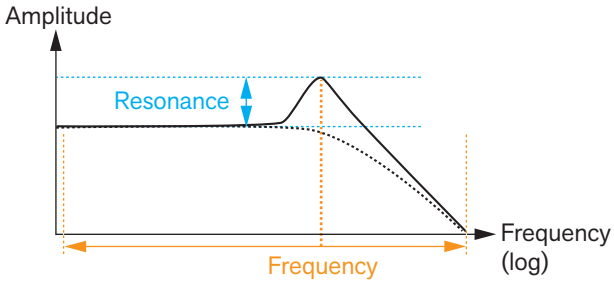
- **MFB HP 24dB**



A multiple feedback (MFB) highpass filter with a 24dB/octave slope. If you turn up the Resonance high, additional resonance peaks appear.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

- **K35 LP 12dB**



An "early MS-20 type" of lowpass filter with a 12dB/octave slope. The filter can be driven to self-oscillate.

! **Be careful when using this filter type as high Resonance values could generate quite extreme audio levels!**

LFO



The LFO can be used for cyclic modulation of the Frequency parameter of the Phaser/Flanger/Filter section - and/or for modulating the Volume. The LFO Rate can also be synced to the Reason sequencer. You can also modulate the LFO Rate from the Modulator (see "The Envelope Modulator" and "The Audio Follower Modulator").

Waveform selector



→ **Click the up/down triangles - or drag the waveform display up/down - to select the desired LFO waveform.** Ten different LFO waveforms are available. Besides the standard waveforms (sine, triangle, pulse, etc.) there are random, slope and stepped waveforms. The shape of the waveforms are shown in the display.

! **Note that all waveforms except the "Decay" are bipolar, i.e., they generate both positive and negative levels.**

Rate



Here you set the LFO Rate.

Range: 0.050-50.00 Hz

→ **Click the SYNC button to sync the LFO Rate to the main sequencer tempo.**
Range in Sync mode: 8 Bars to 1/64.

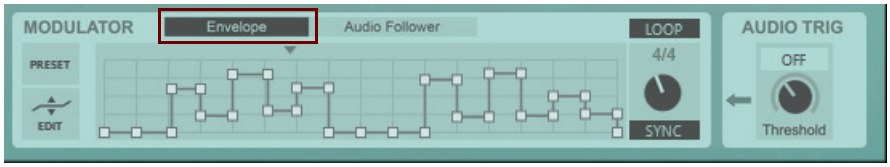
Rate Mod



If you like, you can modulate the LFO Rate from the Modulator signal (see "The Envelope Modulator" below). If the LFO is in SYNC mode, modulating the Rate will force the LFO to switch between the sync divisions.

→ **Set the desired LFO Rate Modulation amount with the knob.**

The Envelope Modulator



The Modulator section features an Envelope and an Audio Follower. You can use either the Envelope or the Audio Follower (but not together).

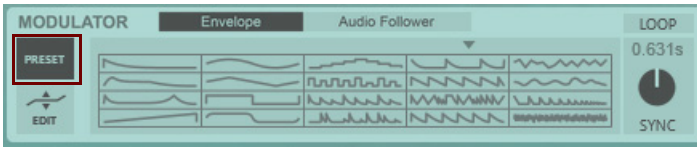
The Envelope is taken straight from the Europa Shapeshifting Synthesizer, so if you are familiar with Europa you will find your way around easily. The Envelope is extremely flexible, and you can draw your own custom modulation shapes by clicking and drawing in the display area. There are also a number of preset shapes that you can use as starting points (or use as is). If you use Loop mode, you could turn the envelope into an advanced LFO and design your own wave shapes.

The Envelope can then be used for modulating the Frequency parameter of the Phaser/Flanger/Filter section, for modulating the Volume, and for modulating the LFO Rate.

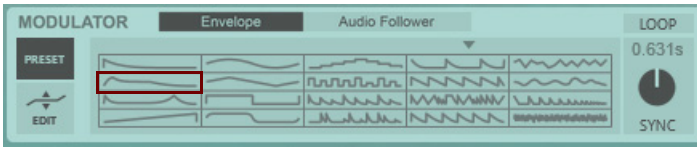
- **The envelope/loop playback starts as soon as there is audio present in Sweeper - or you can trigger playback using the Audio Trig function (see "Audio Trig") or a CV trig signal on the rear panel (see "Trig Envelope") instead.**

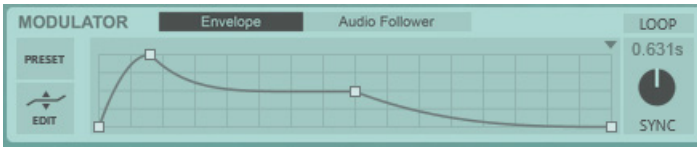
Preset

1. **Click the Preset button to bring up a palette of envelope preset curves:**



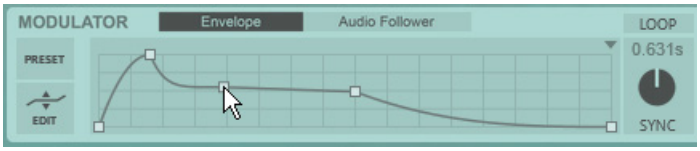
2. **Click the desired envelope preset curve to place it on the display.**
 Let's select a standard ADSR style of envelope curve:





Adding and removing envelope points

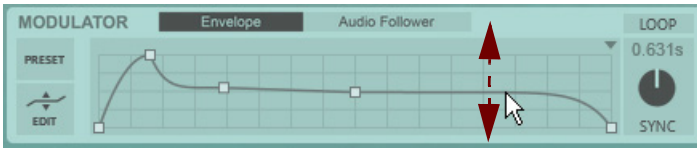
- **Double click, or hold down [Ctrl](win) or [Cmd](Mac) and click in the envelope display to add points to the envelope curve:**



- **To remove a point, double click, or hold down [Ctrl](win) or [Cmd](Mac) and click, on an existing point on the envelope curve.**

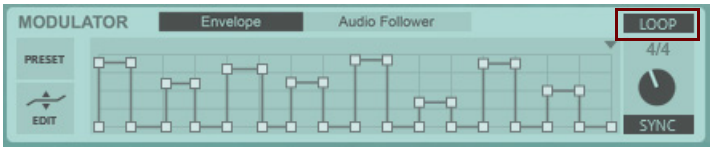
Changing the envelope curve shape

- **Click a line segment (between two points) and drag up/down to change the curve shape:**



Looping the envelope

→ **Click the Loop button to turn the envelope into a kind of LFO.**

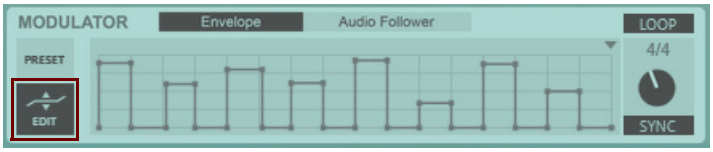


Here we have edited a stepped curve from the Presets. We have also enabled Sync and set the rate to 4/4. This means that each step in the curve now represents an 1/16th note.

- **The loop playback is synced to the Reason sequencer and time line (so that the loop always start on the "one") and continues for as long as there is still audio present through the Sweeper device.**

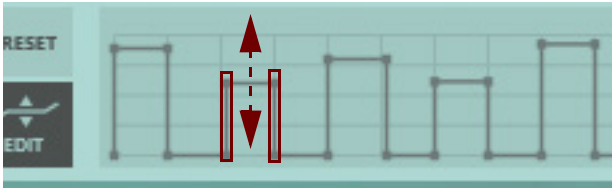
Editing levels only

→ **To restrict the editing to levels only, without affecting the time positions, click the Edit button:**



In this mode you cannot change the time positions of the envelope points, only their levels (height). This is extra useful with a stepped Preset curve, because dragging up or down will change the value of an entire segment, turning the Envelope into a pseudo-sequencer.

! **To be able to adjust the level of a segment, the two points on either side of the segment have to be on the exact same time positions. Otherwise, only the closest point will be changed. Also, any inclining/declining segment will automatically turn horizontal when edited:**

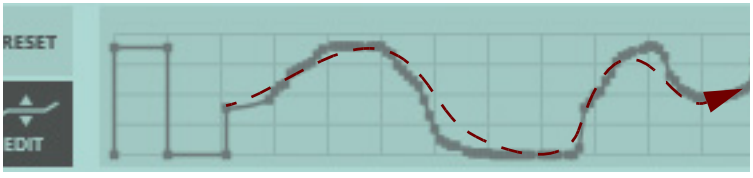


Adjusting the level of a segment.

Creating "free form" envelope curves

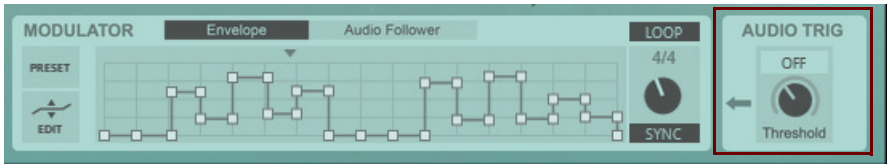
In the Edit mode, you can also draw "free form" curves:

→ **To continuously add new consecutive points, hold down [Ctrl](win) or [Cmd](Mac) and drag in the envelope display:**



→ **To erase points, hold down [Shift] and [Ctrl](win) or [Cmd](Mac) and drag in the envelope display.**

Audio Trig



It's also possible to trigger the envelope from the audio running through Sweeper.

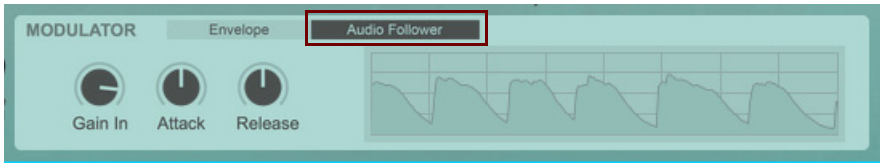
→ **Activate the Audio Trig function and set the Threshold value as desired.**



When the audio level in Sweeper exceeds the set Threshold value, or quickly increases when above the Threshold level, the envelope is triggered.

- **If the envelope is not in Loop mode, one complete cycle is completed at the most. The cycle continues to play back as long as there is audio present through the Sweeper device.**
- **If the envelope is in Loop mode, the loop plays back from the very beginning when an Audio Trig signal is received. The loop continues to play back as long as there is audio present through the Sweeper device.**

The Audio Follower Modulator



The other part of the Modulator section is the Audio Follower. This is basically an envelope follower, which tracks the level of the audio running through Sweeper and outputs a control signal that can be used for modulating the Frequency parameter of the Phaser/Flanger/Filter section, for modulating the Volume, and for modulating the LFO Rate. The tracked (followed) audio level is shown in real-time in the display.

Gain In

Here you can attenuate or gain the modulation signal level, to adjust it to the audio signal level.

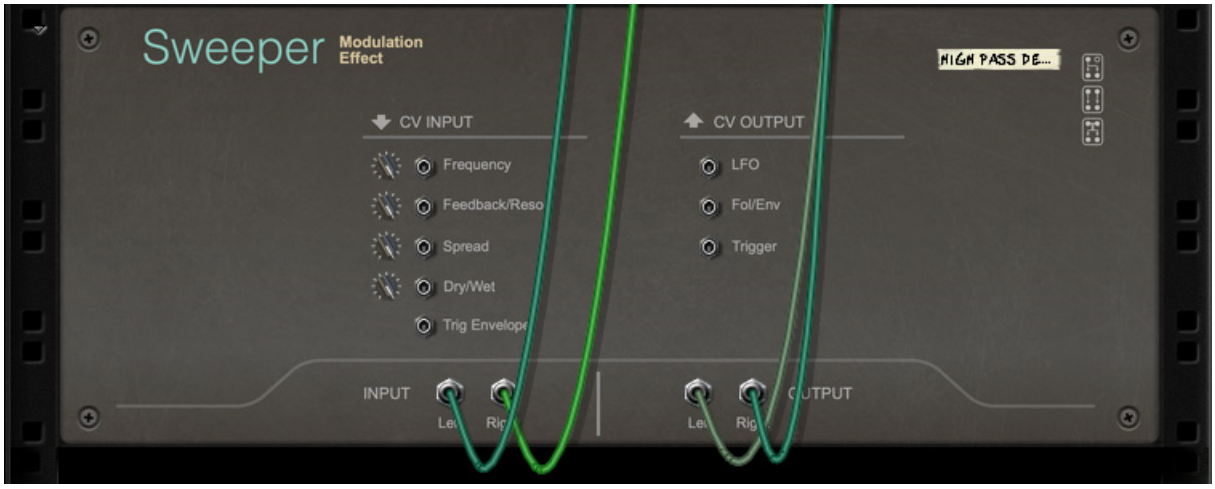
Attack

This controls how fast the envelope follower should react after the input signal level has increased from one value to a higher.

Release

This controls how fast the envelope follower should react after the input signal level has decreased from one value to a lower.

Connections



CV Input

Frequency

This CV input can be used for modulating the Frequency parameter in the Phaser/Flanger/Filter. The input accepts bipolar control signals. The input signal can be attenuated with the corresponding attenuation knob.

Feedback/Reso

This CV input can be used for modulating the Feedback or Resonance parameter in the Phaser/Flanger/Filter. The input accepts bipolar control signals. The input signal can be attenuated with the corresponding attenuation knob.

Spread

This CV input can be used for modulating the Spread parameter in the Phaser/Flanger/Filter. The input accepts bipolar control signals. The input signal can be attenuated with the corresponding attenuation knob.

Dry/Wet

This CV input can be used for modulating the Dry/Wet parameter in the Phaser/Flanger/Filter. The input accepts bipolar control signals. The input signal can be attenuated with the corresponding attenuation knob.

Trig Envelope

This CV input can be used for triggering the Envelope Modulator. The Envelope Modulator is triggered as soon as the CV value goes from zero to a positive value.

If the Audio Trig function is active (see "Audio Trig"), the Trig Envelope function will co-exist with this.

CV Output

LFO

This sends out the LFO signal as a bipolar CV signal.

Fol/Env

This sends out the Audio Follower or Envelope signal as a unipolar CV signal.

Trigger

This sends out a unipolar CV trig signal as soon as the Audio Trig function (see "Audio Trig") is triggered.

Input Left & Right

→ **Patch the audio signals you want to process here.**
If your input signal is in mono, connect only to the L (left) input.

! **Note that the result also depends on the current Routing setting (see "Routing").**

Output Left & Right

These are the stereo audio outputs.

! **Note that the result also depends on the current Routing setting (see "Routing").**

Chapter 31
Alligator
Triple Filtered Gate

Introduction



The Alligator is a three-channel gate effect with a built-in pattern player. It can chop up audio in a wide variety of ways and process it with three parallel filters, distortions, a phaser and a delay. The Alligator can be used for processing sustaining sounds like strings and pads, adding rhythms and accents. It can also be used on loops and other rhythmic material, changing the feel and sound. Applied to a whole mix, the Alligator can be a powerful remix tool, totally reshaping the material.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

About the Patch format

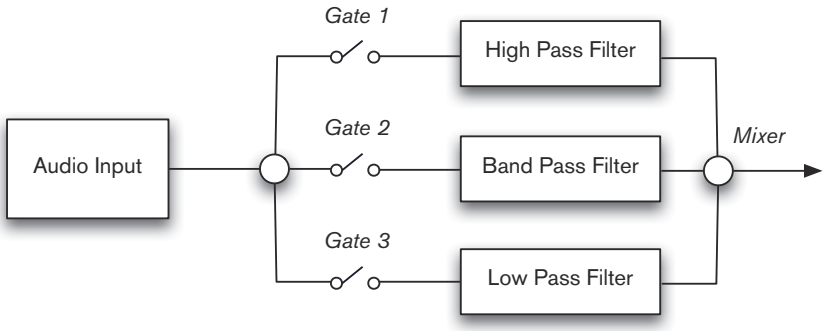
Alligator patches have the file extension ".gator". The Factory Sound Bank contains a number of Alligator effect patches for use as is or as starting points for further tweaking. Patches are loaded and saved using the standard procedures.

► **Don't forget that you can also save Alligator settings as part of a Combinator patch. Combining an instrument device with an Alligator is a quick way to create gated, rhythmic pads.**

Overview and signal flow

The Alligator may seem overwhelming at first - it's got quite a few knobs and buttons on its front panel. However, once you've understood the basic signal flow it's actually pretty straightforward. Read through the description below and get familiar with the basics - it will help you a lot when working with the Alligator.

Here is a simplified diagram of how the Alligator works:



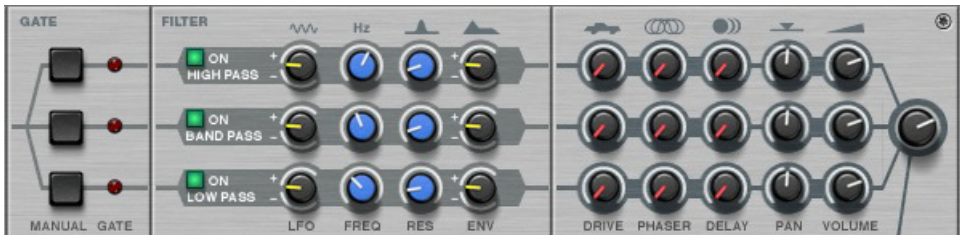
You normally connect the Alligator as an insert effect, so that all of the audio signal passes through the effect device. The incoming signal is split into three, parallel channels. For each channel, there is a separate gate - when that gate is open the signal passes through and when it's closed, the channel is silent. The gates can be opened in four ways:

- **By the built-in patterns.**
 There are 64 patterns, each with three "tracks" independently controlling the three gates.
- **By clicking the Manual Gate buttons on the front panel.**
- **By triggering the gates with the MIDI notes F#1, G#1 and A#1.**
 This way you can play the Alligator live, with velocity control over the gate levels, see "Playing the Alligator live".
- **By connecting CV cables to the Gate inputs on the back of the Alligator and sending Gate signals, e.g. from a Matrix or Redrum.**

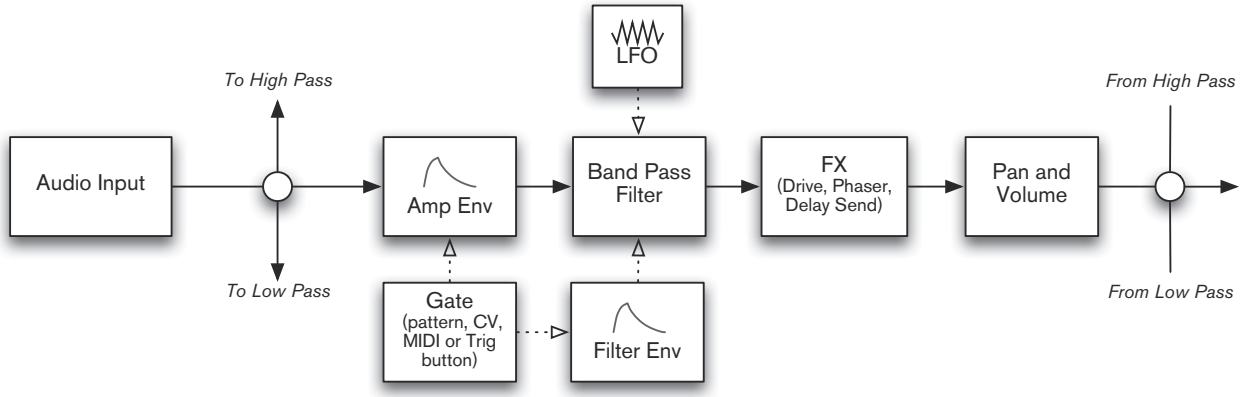
When a gate is open, the signal passes through a filter. The three channels have different types of filters: High Pass, Band Pass and Low Pass, respectively. This means the channels will have different sound characteristics.

Finally, the three channels are mixed together again and sent to the main output.

That's the signal flow in its most basic form. Looking at the front panel, you can see the signal split and the three channels with their gates and filters:



However, as you can see, there are quite a few other settings as well. Let's take a closer look at one of the channels (the band pass filter, in this example):



In this, more detailed diagram, we see that the gate isn't a simple on/off switch - there is actually an amplitude envelope controlling the volume of the channel. When the gate is opened, the envelope is triggered and the sound is let through according to the envelope settings. You can use the amp envelope to soften the attack, to make the notes shorter and more snappy, etc. The gate also triggers a filter envelope, so that each note can get an articulated filter contour. The filter can also be modulated by a global LFO.

Next in the channel are FX settings: a distortion unit, a swirling phaser and a send to a built-in delay unit. Since these settings are independent for the three channels, they can give you a lot of variations.

Finally, there are Pan and Volume controls. Even a function as basic as stereo panning can make for really interesting, spatial effects - especially since you can pan the three channels, the dry signal and the delay independently!

Parameters

Common effect device parameters

Like all effect devices, Alligator features a Bypass/On/Off switch and an input level meter. These are described in "Common effect device features".

Pattern section



Pattern On

When this is on, the built-in pattern player will run in sync with the song tempo, controlling the three gates. Turn it off if you want to control the gates manually or with MIDI/CV.

Shuffle

Shuffle on the Alligator works in the same way as shuffle on the Redrum and Matrix devices. It will delay every second 1/16th note in the playing pattern according to the Shuffle amount setting in the I/O device, creating a shuffle or swing feel.

Note that Shuffle will work best when Resolution is set to 1/16.

Pattern selector

This is where you select which one of the 64 built-in patterns should play back, controlling the gates. There is a guide to the patterns in "The built-in patterns".

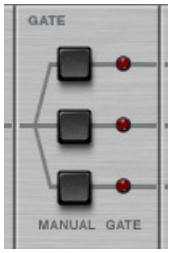
Resolution

When this is set to 1/16 (default) the built-in patterns will be based on 1/16th notes. Changing the Resolution setting allows you to scale the patterns, making them play back faster or slower in relation to the song tempo.

Shift

This will offset the pattern relative to the song playback, moving it "sideways". The range is ±16 steps, with the step length determined by the Resolution parameter. For example, if you set Shift to -1 with Resolution at 1/16, the pattern will be moved one sixteenth note to the left. This means the pattern will play one sixteenth note "early" (the start of the pattern will occur a sixteenth note before the downbeat in the song).

Gate and Amp Envelope



Manual Gate Trig buttons

Clicking one of the Manual Trig buttons will open the corresponding gate. It will remain open for as long as you keep the mouse button pressed. However, the sound may fade out slowly or quickly depending on the amplitude envelope settings.

While the gate is held open manually, it won't be affected by the built-in pattern. This means you can use the buttons to override the pattern, holding a channel open (if Amp Env Decay is long) or muting it (if Amp Env Decay is short).

Gate indicators

These light up when the gates are open.

Amp Env Attack



When a gate is opened, the Amplitude Envelope is triggered. This controls the input level to the corresponding filter. Amp Env Attack sets how long it takes for the level to reach its maximum after the gate opens. Normally, this is kept at a low value for quick, snappy attacks. Raising the Attack parameter will make the notes fade in, blurring the patterns.

Amp Env Decay



Directly after the attack phase, the input level will fade down to zero again. The time this takes is set with the Amp Env Decay parameter. Setting the Decay knob to its maximum value will set the decay time to infinity, which will result in a maximum "sustain" level. Lowering the Decay setting will make the pattern notes shorter.

Amp Env Release



This determines how quickly the sound fades out after the gate is closed. If you raise this setting, the sound will never fade out completely between gates, and the pattern will become blurred and more pad-like.

Filters and Modulation



The three channels have identical settings, even though their filters are of different types. Below, all descriptions apply to all three channels, if not explicitly stated.

Filter On button

When this is on, the channel's signal passes through the filter. Turning this off bypasses the filter. Note though that the Gate, Amp Envelope, effects and other settings are still active.

LFO Amount

Determines how the filter frequency should be affected by the global LFO (see below). This is a bipolar control, allowing for positive or negative modulation of the filter frequency.

Frequency

► **For a general introduction to different filter types, see "The Filter Section" in the Subtractor chapter.**

- **For the high pass filter, this is the cutoff frequency.**
 Frequencies below this will be removed from the signal. Turning this parameter up will gradually remove more and more of the signal, leaving only the highest frequencies.
- **For the band pass filter, this is the center frequency.**
 Lower and higher frequencies will be removed from the signal.
- **For the low pass filter, this is the cutoff frequency.**
 Frequencies above this will be removed from the signal. Turning this parameter down will gradually remove more and more of the signal, leaving only the lowest bass frequencies.

Resonance

The filter resonance emphasizes the frequencies around the set filter frequency. Turning this up will make the filter sound more pronounced and ringing.

Envelope Amount

This determines how the filter frequency is affected by the Filter Envelope (see below). This is a bipolar control, allowing for positive or negative modulation of the filter frequency.

LFO Waveform



The global LFO offers nine different waveforms, ranging from sine, triangle and square to random and various stepped forms.

LFO Frequency



Sets the rate of the LFO, used for continuous modulation of the filters. If LFO Sync is activated, the LFO Frequency is expressed as a note value relative to the song tempo; if not, the LFO Frequency is free.

LFO Sync



Turn this on to synchronize the LFO to the song tempo.

Filter Env Attack



Like the amplitude envelope, the filter envelope is triggered by the gates. There are in fact three individual envelopes, one for each filter, but they share the same controls. For the filter envelopes to have any effect on the sound, you need to set the Env Amount parameters to negative or positive values for one or more filter channels.

The Filter Env Attack determines how quickly the filter envelope rises to its maximum value when the gate is opened.

Filter Env Decay



Directly after the attack phase, the filter envelope signal will fall to zero again. The time this takes is set with the Filter Env Decay parameter.

Filter Env Release



This determines how quickly the filter envelope signal falls to zero after the gate is closed. To fully hear the effect of this parameter, you need to raise the Amp Env Release parameter - otherwise the level will drop to zero directly when the gate closes and you won't hear any filter changes.

Effects



The three channels have identical effect parameters. Distortion and phaser effects are separate for the three channels (although the phasers have common controls). The delay is a global effect, working much like a send effect in a mixer.

Drive Amount



Sets the amount of distortion for the channel.

Phaser Amount



Sets the amount of phaser effect for the channel.

Delay Amount



This works like an effect send, determining how much of the signal should be sent to the built-in delay effect. The send is post-volume: If you lower the volume for a channel, the signal sent to the delay will be lowered as well.

Delay Time



This is a standard delay unit with a maximum delay time of 2/4 (when synced to the song tempo) or 1 second.

Delay Sync



Turn this on to set the delay time in musical values relative to the song tempo.

Delay Feedback



This determines the number of delay repeats.

Delay Pan



Sets the stereo panning of the delay repeats.

Phaser Rate



The rate of the phaser sweep.

Phaser Feedback



This is similar to the resonance control on a filter. Raise the feedback to get a more pronounced, “singing” phaser effect.

Mix controls



These parameters determine the signal mix being sent to the main outputs on the back. There are also individual outputs for the three gate/filter channels. If you connect these outputs, the corresponding channel signals will be removed from the main mix, leaving only the delay return signal and the dry signal.

Channel Pan



Sets the stereo pan/balance of the channel.

Channel Volume



The volume of the channel.

Dry Ducking



The Ducking parameter will apply the Amp Envelope to the dry signal - but inverted. This means that whenever the Amp Envelope is “high”, the dry signal will be lowered in volume or “ducked”. The result is a sort of mirror to the sound from the three gated channels.

! **Note that this is only audible if the Dry Volume has been raised.**

Dry Pan



Sets the stereo pan/balance of the dry, unprocessed signal.

Dry Volume



Sets the volume of the dry, unprocessed signal. Mixing in a bit of the dry sound is useful for subtler processing, e.g. when you just want to animate a pad rather than chop it up.

Master Volume



This is the master volume of the mixed signals. The signals from the separate channel outputs on the back won't be affected by this.

Audio connections



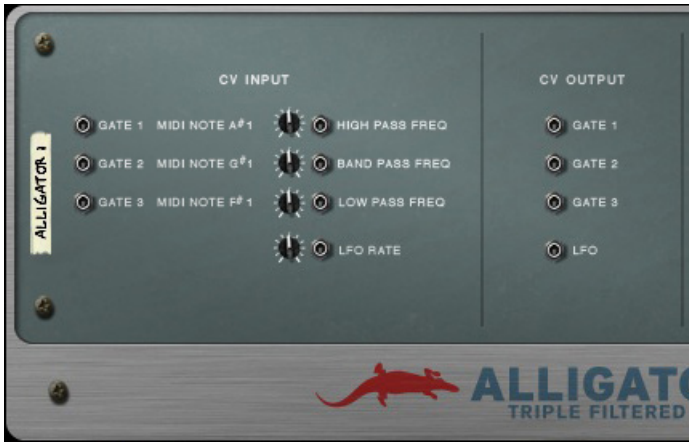
Main Inputs and Outputs

The Alligator is normally connected as a stereo in-stereo out effect. Should you connect a mono input signal, the output will still be in stereo due to the pan controls.

Separate Outputs

These output the signals from the individual gate/filter channels. Connecting one of these outputs will remove the corresponding channel signal from the main output. The separate output signals are taken after the Channel Volume controls but are unaffected by the Master Volume.

CV connections



Gate inputs

These are used for controlling the gates from other devices, using CV. When a gate input receives a CV value of 7 or higher, the gate will be opened. Higher values result in higher input level for the gate channel (i.e. the gates are velocity sensitive).

- **If you select an Alligator device and create a Matrix Pattern Sequencer, its gate output will be auto-routed to the first available Gate input. Also, the Matrix Curve CV output will be auto-routed to the corresponding CV Freq input on the Alligator.**
 You can create up to three Matrix Pattern Sequencers with the Alligator selected and the Matrix devices will be auto-routed to separate Gate and CV Freq inputs on the Alligator.

! **Note: If you want the gates to be controlled by CV only, you need to turn off the Pattern player on the front panel. Otherwise, the gate sources will be combined.**

CV Modulation inputs

These jacks allow you to modulate the filter frequencies of the three filters, as well as the global LFO rate.

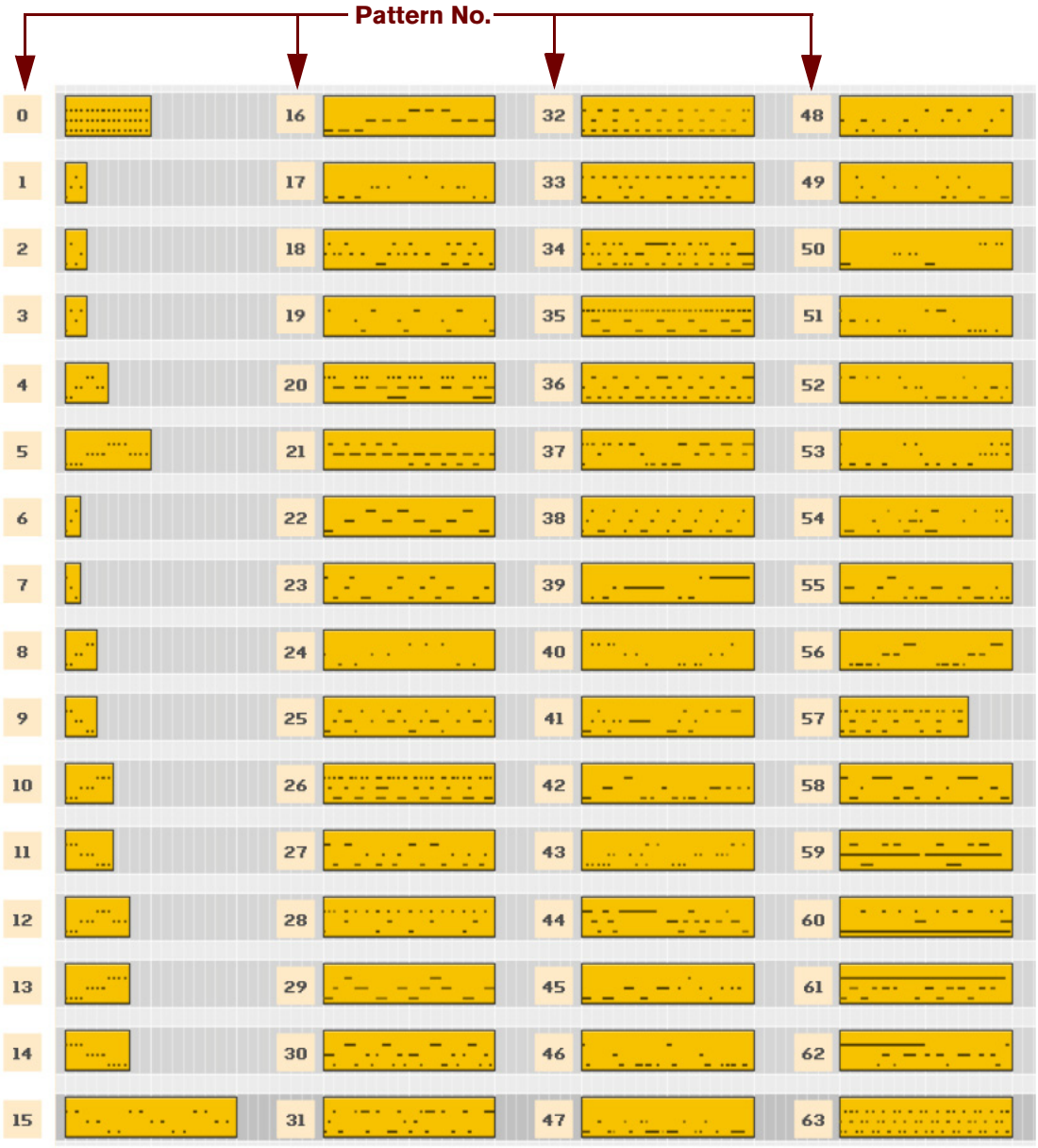
Gate Outputs

The three Gate outputs simply send out the current Gate values, regardless of whether these are controlled by the built-in pattern player, the buttons on the front panel, MIDI or CV. You can use these to trigger other sounds and effects in time with the gates.

LFO CV Out

This is the output of the built-in LFO, for modulating parameters in other devices.

The built-in patterns



This is an overview of the 64 built-in patterns. The black dots signify open gates with gate 1 (high pass filter) at the top. Most patterns are two bars long, but some are shorter. All patterns will repeat continuously when the Pattern function is on.

Methods and Tips

Playing the Alligator live

The three gates in the Alligator can be triggered by the MIDI notes F#1, G#1 and A#1, with the lowest note controlling the low pass filter channel and so on. This can be very useful when playing live:

1. **Create and connect an Alligator as an insert effect for the audio material you want to process.**
 This could be a recorded pad, a loop or even the full mix.
2. **Make sure the Alligator is selected and select "Create Track for Alligator" from the Edit menu.**
 This creates a track with a note lane for the Alligator.

→ **If you like, turn off the Pattern section on the Alligator.**

3. **Start playback and use the keys F#1, G#1 and A#1 to play the gates.**
 Remember that the gates are velocity sensitive and that the Amp Envelope settings affect the gated sound.

Playing the gates from Matrix patterns

While the 64 built-in patterns are rather versatile, sometimes you might want to create your very own patterns. The easiest way to achieve this is to connect Matrix Pattern Sequencers:

1. **Select the Alligator device.**
2. **Create a Matrix from the Create menu or Device palette.**
 It is automatically connected to one of the Gate inputs on the backside. The Curve CV output on the Matrix is automatically connected to the corresponding CV Freq input on the Alligator.
3. **Repeat these two steps twice, to create two more Matrix devices.**
 Note that you need to select the Alligator for the Matrix to be auto-routed to the Gate and CV Freq inputs. Now you have three Matrix devices, one for each gate/channel.
4. **Turn off the Pattern player in the Alligator.**
5. **Use the Gate section in the Matrix display to program custom gate patterns.**
 The Curve can be used for controlling the Filter Frequency of the corresponding frequency band in the Alligator.

► **Another interesting trick is to load a rhythmic loop into a Dr Octo Rex device and connect its Slice Gate Output to a Gate input on an Alligator. This makes the slices in the REX loop trigger the gate.**

Controlling other sounds and effects

The gate outputs on the Alligator back panel allow you to control other stuff with the built-in Alligator patterns. Here's an example:

1. **Create an instrument with a sustaining sound, such as a pad.**
2. **Turn on loop mode in the main sequencer and record some chords.**
3. **Create an Alligator and select a suitable pattern.**
 You might want to tweak the sound as well, change the filter settings, adjust fx and pan, etc.

With some tweaking, the filtered pattern can sound a bit like a drum beat or percussion loop. Then it can be nice to add some punch and bottom to the Low Pass channel, making it feel more like a kick drum or bass line:

4. **Create a Kong Drum Designer.**
5. **Load a punchy bass drum sound into Pad 1.**
6. **Flip the rack around and route a CV cable from Gate 3 Out on the Alligator to the Gate input for Kong's pad 1.**
 Now the bass drum sound is triggered along with Gate 3, played by the Alligator's pattern player.
7. **Adjust the Kong level to taste.**
8. **If you like, add other sounds for Gate 1 and 2 in the same way.**



Chapter 32
Pulveriser

Introduction



Pulveriser is a very versatile stereo in/out compression+distortion+filter device, capable of mangling any sound literally beyond recognition, but also capable of producing more subtle musical effects. Pulveriser features a wonderful compressor, coupled with a nice warm distortion, plus a multi-mode filter. The different sections of Pulveriser can be modulated by an LFO and by an Envelope Follower to allow for really organic modulation effects. Pulveriser also features a Dry/Wet mix control so you can utilize parallel processing in the unit itself, i.e. mix in the processed signal in parallel with the dry signal - great for parallel compression of drum loops etc.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

About the Patch format

Pulveriser features programmable effect presets. Included are a number of factory Patches which can be used as they are or provide you with a good starting point for further tweaking.

Patches use the file extension "*.pulver". Loading and saving Patches is done in the same way as for instrument devices.

Parameters

Pulveriser contains three main effect sections; Squash (compression), Dirt (distortion) and Filter (multi-mode filter with cutoff and resonance controls). In addition, Pulveriser features two modulation sections - an LFO with selectable waveforms and an Envelope Follower which can modulate the effect sections. The parameters in each section are as follows:

Common effect device parameters

While the specific parameters for the Pulveriser effect device are described below, some features and procedures are common to all effect devices. Please, refer to "Common effect device features" for information about the Bypass/On/Off switch.

Note, however, that the Input Meter is different on the Pulveriser device. Instead of the standard multi-segment LED indicator found in other Reason effect devices, Pulveriser features a red lamp which reflects the intensity of the input signal.

Signal Routing selector



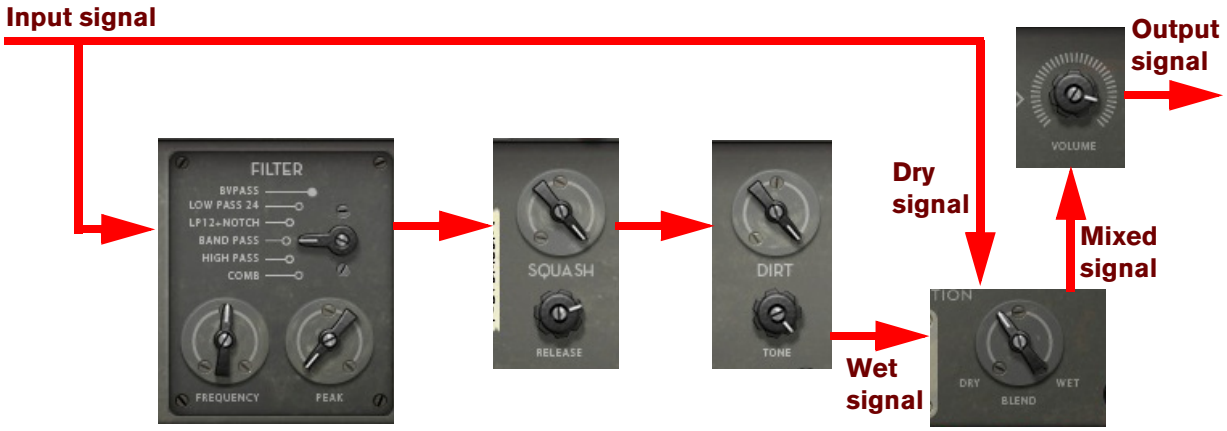
With the Signal Routing selector to the bottom left on the front panel you determine the signal flow through Pulveriser:

- **Squash > Dirt > Filter**
 This setting puts the compressor before the distortion and filter sections in series:



 In this configuration the compressor and distortion affect the entire unfiltered input signal.

- **Filter > Squash > Dirt**
 This setting puts the filter before the compressor and distortion sections in series:



 In this configuration the compressor and distortion affect the filtered input signal. This way you can define what frequencies should enter the compressor and distortion sections.

The Squash section



The Squash section is where you set the compression of the signal - from subtle compression to heavily “pumping” effects.

Squash

The Squash knob affects compression ratio, threshold and make up gain in a nice and musical way.

Release

The Release knob controls the compressor release time. If you set this to a fairly low value and have the Squash amount fairly high, you will get that nice “pumping” compression effect.

The Dirt section



The Dirt section is where you can add distortion to the signal - from gentle to screaming outrage.

Dirt

The Dirt knob controls the level of distortion.

Tone

The Tone knob controls a lowpass filter. Turned fully clockwise the filter is completely open, letting all frequencies through.

The Filter section



The Filter section features five different filter types. Each filter type has controls for Frequency and Peak (resonance amount).

Filter selector

Select one of the following different filter types (or bypass):

- **Bypass**
 This will let the input signal through the Filter section unaffected.
- **Lowpass 24**
 This is a lowpass filter with a slope of 24dB/octave, similar to the LP24 lowpass filter in the Subtractor (see "24 dB Lowpass (LP 24)" in the Subtractor chapter).
- **LP12+Notch**
 This is a lowpass filter with a slope of 12dB/octave, in series with a notch filter. The cutoff frequency of the low-pass filter and the center frequency of the notch filter are the same. Changing the Frequency control will generate sort of an "animated" effect.
- **Band Pass**
 This is a bandpass filter similar to the BP12 filter in the Subtractor (see "Bandpass (BP 12)" in the Subtractor chapter.
- **High Pass**
 This is a highpass filter with a slope of 12dB/octave, similar to the HP12 filter in the Subtractor (see "Highpass (HP12)" in the Subtractor chapter.
- **Comb**
 This is a comb filter similar to the "Comb +" Filter in Malström (see "Comb + & Comb –" in the Malström chapter).

Frequency

The Frequency knob controls the cutoff frequency or center frequency depending on selected filter type.

Peak

The Peak knob controls the resonance amount.

The Tremor section



The Tremor section in Pulveriser is an LFO which can modulate the Filter Frequency parameter and/or the main output Volume parameter. The Tremor section features the following parameters:

Rate

Controls the LFO rate. The Rate range in Pulveriser is very wide and can reach way up in the audio frequency range. The rate can also be synced to the sequencer tempo by clicking the Sync button to the right (see "Sync" below). In sync mode, the Rate knob controls the sync resolution.

As a special feature, the rate can also be modulated from the envelope follower in the Follower section, see "The Follower section".

Waveform selector

Select one of nine LFO waveforms. Apart from standard waveforms (sine, triangle, square etc.) there are random, non-linear and stepped waveforms. The shape of the waveforms are shown in the display and reflect how the modulated signal is affected.

Sync

Click the Sync button to synchronize the LFO Rate to the main sequencer tempo. In Sync mode the LFO cycle is also synced to the sequencer, which means that the LFO cycle chases the current sequencer position.

Control the sync resolution with the Rate knob, see "Rate" above.

Spread

The Spread button introduces a stereo effect by modulating the left and right audio channels with two LFO signals phase shifted 180 degrees in relation to each other. This means that you are able to generate kind of a "roto-speaker" effect to the processed signal.

Lag

The Lag control acts like a lowpass filter on the LFO signal, making the LFO signal smoother. This is especially noticeable on waveforms with sharp edges or transients like the square, sawtooth and stepped waves. On the sinewave you will barely notice any effect since it's already "smooth" by nature.

Modulation Amount knobs

On either side of the Tremor section are two bipolar modulation amount knobs, with zero modulation at the 12 o'clock position. Since the modulation parameters are bipolar it means that if the knob is in the "-" sector it will invert the LFO wave shape. This is especially useful in LFO Sync mode where you can define the "direction" of the modulation in relation to the sequencer playhead.

- → **Turn the left knob to control the modulation of the Frequency parameter of the Filter section.**
- → **Turn the right knob to control the modulation amount of the wet signal of the Volume parameter.** This will introduce a tremolo effect to the Wet signal fed via the Volume control.
- ! **Note that the Dry signal in the Blend section is never affected by the tremolo effect! This means that if the Blend knob is set to fully Dry, there won't be any tremolo effect of the volume.**

The Follower section



The Follower section features an envelope follower which analyzes the amplitude of the incoming signal and outputs a modulation (CV) signal that corresponds to the incoming audio level. The modulation signal can then control the Frequency parameter in the Filter section and/or the LFO Rate parameter in the Tremor section. The Follower section features the following parameters:

Trig

Click/hold the Trig button to manually trig/gate the envelope follower. Clicking/holding the Trig button will make the envelope follower output a modulation signal according to the settings of the Attack and Release parameters described below. If you hold the Trig button for a longer period than the Attack time, the Follower will output maximum CV signal level. When you then release the Trig button, the CV signal level will drop according to the Release time and continue to follow the audio input signal level instead.

Threshold

This defines at which input signal level the envelope follower should trig. Set to a low value, the envelope follower will react as soon as there is any audio signal present on the Pulveriser inputs. Set to a high value, the envelope follower will react only on loud input signals, or from a manual Trig signal.

The red lamp to the right of the Threshold knob gives a visual indication of the CV signal level.

- **On the back of Pulveriser you will find a Follower CV output - this delivers the CV signal from the envelope follower, allowing you to dynamically control parameters in other devices.**

Attack

This controls how fast the envelope follower should react after the input signal has reached above the Threshold value. Note that the attack time can only be increased compared to the input signal - never shortened.

Release

This controls how fast the envelope follower CV signal should drop to zero after the input signal has decreased below the Threshold value. Note that the release time can only be increased - never shortened.

Modulation Amount knobs

To the left of the Follower section are two bipolar modulation amount knobs.

→ **Turn the upper left knob to control the modulation of the Rate parameter of the Tremor section.**
If the modulation knob is in the "+" sector the rate will increase according to increased audio level. If the modulation knob is in the "-" sector the rate will decrease according to increased audio level.

If Sync is enabled in the Tremor section, the rate will jump between the different resolutions according to the Follower modulation amount.

! **Modulating the Tremor Rate in Sync mode can produce noise when the Rate parameter switches between resolutions. This noise can be eliminated by increasing the Lag amount in the Tremor section, see "Lag".**

→ **Turn the lower left knob to control the modulation amount of the Frequency parameter of the Filter section.**
If the modulation knob is in the "+" sector the Filter Frequency will raise according to increased audio level. If the modulation knob is in the "-" sector Filter Frequency will drop according to increased audio level.

Blend



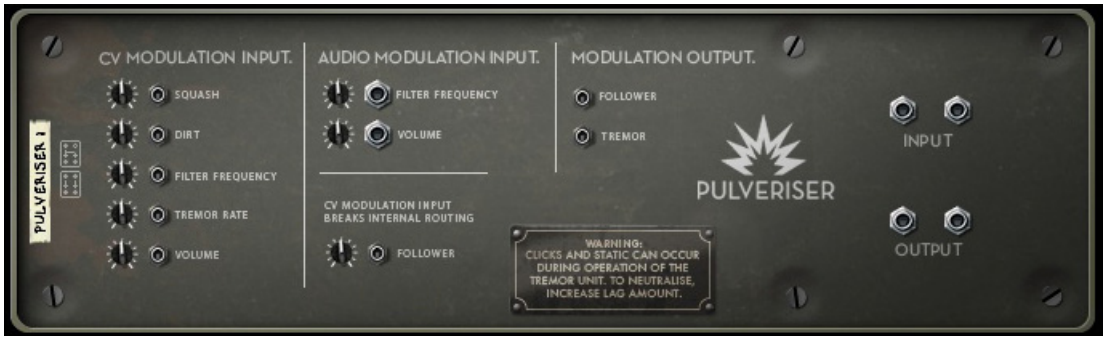
With the Blend knob you control the mix between the dry and wet signal. With the knob set somewhere in between the Dry and Wet position you will have parallel processing. This can be useful if you, for example, want to process a drum loop with compression (Squash) and distortion (Dirt) and mix the processed signal with the dry before sending it to the outputs.

Volume



With the Volume knob you set the total output level of the dry+wet signals.

Modulation inputs and outputs



CV Modulation inputs

On the back of Pulveriser you will find CV inputs for controlling the following parameters:

Squash

Use this for dynamically changing the amount of compression in the Squash section.

Dirt

Use this for dynamically changing the amount of distortion in the Dirt section.

Filter Frequency

Use this for dynamically changing the Frequency parameter in the Filter section.

Tremor Rate

Use this for dynamically changing the LFO Rate parameter in the Tremor section.

If the LFO is in Sync mode, the rate will jump between the different resolutions according to the CV modulation input signal amount.

Volume

Use this for dynamically changing the output volume from Pulveriser.

Follower

Use this for controlling the envelope follower signal from an external source. The internal envelope follower signal is replaced with the CV signal that is inserted here.

! **Note that the Attack and Release controls can still be used to shape the CV input signal.**

Audio Modulation inputs

! **Note that these modulation inputs accept audio rate signals, which is really cool!**

Filter Frequency

Use this for dynamically changing the Frequency parameter from an external audio signal. The result of this modulation is Filter FM.

Volume

Use this for dynamically changing the output volume from an external audio signal. The result of this modulation is amplitude modulation (AM) of the Pulveriser output signal.

CV Modulation outputs

Follower

On this output the control signal from the Follower is present.

Tremor

Here, the LFO CV signal from the Tremor section is present.

Demolition tips and tricks

Don't restrict yourself to using Pulveriser as a basic compressor or distortion unit - you may be surprised to find how often Pulveriser can add power, warmth, color and "movement" to your sounds. Here are some examples:

Beef up your sounds

Use Pulveriser to fatten up your sounds:

1. **Select the Low Pass 24 filter type in the Filter section.**
2. **Set the Frequency parameter to a low value so you can barely hear the lowest frequencies.**
3. **Set the Threshold, Attack and Release knobs in the Follower section to minimum values.**
4. **Slowly increase the Follower > Filter Amount knob until you can hear the lower frequencies of the sound.**
5. **Adjust the Blend knob until you are satisfied with the mix.**
6. **Adjust the Blend, Squash and Dirt knobs to fine-tune the sound.**

Make your pads tremble

Use Pulveriser to create movements in your pad sounds:

1. **Select the Comb filter type in the Filter section.**
2. **Set the Peak knob in the Filter section to a fairly low value.**
3. **Turn the Tremor > Filter Modulation amount knob past the 12 o'clock position.**
4. **Set the Tremor Rate knob to around the 10 o'clock position.**
5. **Set the Threshold, Attack and Release knobs in the Follower section to minimum values.**
6. **Set the Follower > Tremor Amount knob to around the 2 o'clock position.**
7. **Set the Tremor > Volume knob to around the 3 o'clock position.**
8. **Play a pad sound which has long Attack, Decay and Release times and a fairly low Sustain value on its amp envelope.**
 Notice how the Tremor Rate and intensity change and modulate the filter and volume according to the pad volume.

Chapter 33
The Echo

Introduction



The Echo is an advanced stereo in/out echo and delay device with a multitude of parameters for tweaking the color and shape of the echo effect - diffusion, filtering, distortion and more. In addition to the normal mode where The Echo behaves like a regular send or insert effect, there are two additional modes called Triggered and Roll which let you automate momentary echo effects as well as create interesting stutter and repeat effects on the fly. The Echo also features breakout jacks which allow you to insert any number of other effect units into the feedback loop - this opens up endless possibilities for creative sculpting of the echo repeats.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

About the Patch format

The Echo features programmable effect presets. Included are a number of Factory Patches which can be used as-is or provide you with a good starting point for further tweaking. The Patches use the file extension "*.echo". Loading and saving Patches is done in the same way as for instrument devices.

Parameters

The Echo is comprised of six main sections; Mode (Normal, Triggered and Roll), Delay (time, tempo sync and stereo behavior parameters), Feedback (including a Diffusion subsection), Color (Drive and resonant Filter), Modulation (Envelope, Wobble and LFO) and Output (Dry/Wet control and Ducking).

Common effect device parameters

While the parameters specific to The Echo are described in detail below, some features and procedures are common to all effect devices. Please refer to "Common effect device features" for further info about the Bypass On/Off switch. It should however be noted that the Input Meter on The Echo differs from the standard 7-segment LED meter found on other effect devices in Reason; The Echo features a 3-segment meter with the usual green, yellow and red LED (the latter indicates clipping).

The Mode section

The Mode section acts as a signal gate with three different ways of passing the input signal on to the subsequent main sections.



Mode

This switch has three positions:

- **Normal**
 The standard send/insert effect behavior where the input signal is fed continuously into the device.
- **Triggered**
 This mode keeps the signal unprocessed until you hit the Trig button. This mode is for situations where you only want the echo effect momentarily, e.g. on every 4th snare hit, or individual words on a vocal track.
- **Roll**
 In Roll mode, the unprocessed signal passes through unaffected until you turn up the Roll slider, see "Roll slider". This gradually suppresses the Dry signal while simultaneously raising the Feedback and mixing the delay/echo output into the Wet signal. The Roll mode is useful for repeat, stutter and glitch effects. For Roll mode, we recommend that the Dry/Wet parameter in the Output section (see below) is set to (or close to) Wet.

► **See "Using the Roll function" for an example of how to use the Roll function.**

Trig

This button is only functional when the Mode switch is set to the Triggered position. When you press the Trig button it opens the Input signal gate, which stays open until you release the button again. Think of it as momentarily enabling an effect send.

Roll slider

This slider is only functional when the Mode switch is set to the Roll position. The Roll slider does three things:

- **Turns up the feedback (internally) to unity (100%) or to the Feedback setting on the front panel, whichever is higher.**
 This starts happening during the slider throw, to catch a good amount of input signal.
- **Closes the input to the delay (you don't want the delay to continue catching the input sound during the roll).**
 This happens late, with a slight delay - meaning it catches a little bit of sound after you've hit the Roll position.
- **Changes the internal mix from dry to wet.**
 This also happens a bit late, so that if you hit a drum beat perfectly with the roll, you will hear that beat (dry) before the delay repeats (roll).

! **Note that the "internal mix" mentioned above is the signal sent to the "Wet" channel in Roll mode. The "Dry" channel still carries the dry signal at all times.**

The key thing here is that you shouldn't set the slider to a value in between - it should go from 0 to full Roll and back. So why do we use a slider and not a switch? Well, things start happening during the slider throw (and actually a bit after you've moved the slider fully to the right).

The slider allows for some sloppiness, making it easier to catch a beat and roll it fully. The result is slightly different depending on how fast you move the Roll slider - in most situations it works best to move it pretty fast and hit Roll exactly on the beat.

- **Although you will probably most often set The Echo to full Wet when Rolling, it might make sense to have a little bit of dry signal leaking through during the roll (making it easier to keep time and return from the roll exactly on the beat). Then you just set Dry/Wet balance to something like 85% Wet, see** "Dry/Wet".
- **See "Using the Roll function" for an example of how to use the Roll function.**

The Delay section



The Delay section features parameters relating to delay time, tempo sync and stereo behavior.

Time

This knob controls the delay time. The delay time range is 1 to 1000 milliseconds. When Sync is enabled, the range is 1/128 notes up to ½ note.

Offset R

This unipolar knob controls the Right channel delay time offset. The higher the Offset R value, the more the Right channel will be delayed in relation to the Left channel - perfect for creating stereo delay effects!

The Offset range is 1 to 1000 milliseconds. When Sync is enabled, the Offset range is "no offset" up to ½ note (according to the same resolution table as for the Time parameter). However, the 1/128 notes offset value is replaced by "no offset".

Keep Pitch

When you manually change the Delay time during recording or playback, you will notice that the pitch of the delay signal also changes. If this effect is undesirable, you can enable Keep Pitch, which will ensure that the pitch remains fixed regardless of changes in Delay time.

Sync

This button enables Tempo Sync, which affects the Time and Offset R parameter.

Ping-Pong

With Ping-Pong enabled, the stereo position of each delay repeat will alternate between left and right. The Pan knob determines the stereo width as well as the position of the initial repeat. When the Pan knob is set to full Left, the first delay bounce will be panned hard Left, the second will be panned hard Right, and so on. When the knob is set to full Right, the order is reversed (R > L > R etc).

The Feedback section



The Feedback section consists of a main section with Feedback and Offset R controls, as well as a Diffusion subsection which allows you to add a kind of 'smearing' effect to the echo.

Feedback

The Feedback knob sets the amount of feedback, i.e. the amount of wet signal fed back into the delay. This determines the number of repeats. At zero feedback there will only be a single repeat. Unity gain is achieved at 100%. If you increase the feedback beyond this it will increase the gain so a distorted signal is produced.

Offset R

This bipolar knob controls the offset in delay Feedback on the right channel. By default, both channels have the same amount of feedback (as determined by the Feedback parameter), but the Offset R knob allows you to add or subtract feedback separately for the Right channel only. The practical result is that you will hear the echo gradually wander from the center to the left or right side.

This control combined with the Offset R in the Delay section can be useful for controlling the length of the effect for the right channel.

Diffusion

Diffusion introduces kind of a "smearing" effect, somewhat reminiscent of diffusion on a reverb. Raising the Amount will introduce additional delay repeats very near the 'original' repeats, and raising the Spread value will spread these repeats out wider.

The Color section



This section features a distortion/limiter and a resonant filter. Each echo repeat is colored before being fed back into the loop, meaning that the distortion and filter effects will be more pronounced with each repeat.

Drive

This knob sets the amount of the selected distortion/limiter effect.

Type

The Type Switch lets you select between 4 different effect algorithms:

- **LIM (Limiter)**
 This produces that typical “compression” effect which you would get from analog limiters.
- **OVDR (Overdrive)**
 This produces an analog-type overdrive distortion effect.
- **DIST (Distortion)**
 Similar to overdrive, but denser.
- **TUBE**
 This emulates tube distortion.

Filter

This is a resonant bandpass filter, hence it lets you create filter effects where either the lower or the higher frequency range is cut (or boosted, in case the Reso control has been turned up).

Freq

This control sets the change in frequency. For each delay repetition, the frequency content will shift in accordance with the frequency setting.

Reso

This knob determines the amount of resonance on the delay repetitions. A different frequency range will be amplified depending on the setting of the Freq parameter.

The Modulation section



This section features parameters for modulating the pitch and stereo image of the echo effect.

Env

The envelope parameter lets you create a kind of bend effect where the pitch of the echo repeats wanders down or up, depending on whether you turn the knob left or right. The knob is bipolar, meaning that there is no Env effect in the default middle (zero) position.

Wobble

This emulates a tape speed wobbling effect where the speed of the “tape” (and consequently, the pitch of the signal) wobbles randomly.

LFO

The LFO subsection modulates the pitch of the left and right channels independently, meaning that it functions as a kind of stereo spread at modest settings and warps the echo completely at heavier settings.

Rate

This knob sets the speed of the LFO.

Amount

This knob sets the amount of LFO.

The Output section



This section is the final output stage where the processed effect signal goes through the standard Dry/Wet control, as well as the optional Ducking control.

Dry/Wet

This is a traditional dry/wet parameter for controlling the relation between the unprocessed and the processed signal. When Roll mode is enabled (see the Mode section for more info), we recommend that the Dry/Wet control is set to Wet only even when The Echo is used as an insert effect.

Ducking

Ducking attenuates the level of the Wet (processed) signal until the amplitude of the Dry signal drops, at which time the Wet signal is faded back in. This is useful for adding a delay effect to the silence that comes after you have played your lovely lead line. The delay will not be heard while you are still playing so you will avoid muddling up the solo.

CV/Gate inputs



On the back of The Echo you will find the following CV inputs:

Trig

This is a Gate input for controlling the Trig function in the Mode section.

Roll

Use this for dynamically changing the Roll amount (corresponding to the Roll slider on the front panel) in the Mode section.

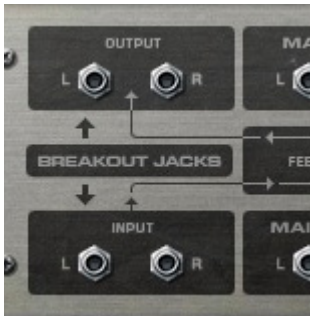
Delay Time

Use this for dynamically changing the Delay Time in the Delay section.

Filter Freq

Use this for dynamically changing the Filter Frequency in the Color section.

The Breakout Jacks



The Echo features special Breakout jacks which allow you to insert other effect devices into the feedback loop. The signal is processed externally and then fed back into the loop, meaning that each delay repeat will be more colored by the effect(s) connected to the breakout jacks.

Tips and Tricks

Using the Roll function

Here's a quick example of how you can typically use the Roll function in The Echo:

1. **Have some signal running through The Echo, e.g. a drum loop.**
2. **Set The Echo to Roll mode and full Wet output (see "Dry/Wet").**
 The Roll slider is at 0. You will hear the drum loop unprocessed, since Roll isn't engaged yet.
3. **On the beat you want to roll (or freeze, to use another term), move the Roll slider quickly all the way to the right.**
 This replaces the dry signal with that beat, being rolled at whatever delay time you set.

- **A nice thing is to set the Delay to Sync mode (see "Sync") so the repeats will be in sync with the loop, e.g. to 1/16th notes or other desired resolution.**

4. **To go back to the (dry) loop, move the Roll slider quickly back to 0.**

Creating "pitched" delay

1. **Set the delay time to minimum (1ms).**
2. **Increase feedback to 100%.**
3. **Enable Diffusion and increase the Amount to maximum.**
4. **Play a drum loop or similar though The Echo.**

Now you have set up a sonic playground that equals no other delay! Experiment with Diffusion Spread, Modulation Env and Delay Time to achieve various interesting sonic results.

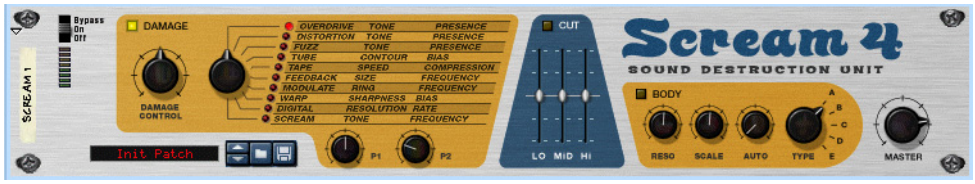
Distorted external feedback

1. **Connect an instrument device to the input(s) of The Echo.**
2. **Connect a Scream 4 device in the feedback loop.**
 Connect the Scream 4 inputs and outputs to the Breakout jacks of The Echo, see "The Breakout Jacks".
3. **Play the instrument device and try different distortion algorithms on the Scream 4.**
 Be careful with the master volume in Scream 4 as this is very sensitive to the feedback level in The Echo.

Chapter 34
Scream 4
Sound Destruction Unit



Scream 4 Sound Destruction Unit



Scream 4 is a very versatile stereo in/out sound destruction device, capable of warping any sound literally beyond recognition, but also capable of producing more subtle musical effects. Scream 4 features a wide range of algorithms for distortion and sound mangling which can be combined with an EQ and a resonant "Body" section to provide everything you need to add an edge to your sounds. This effect is most often used as an insert effect.

About the Patch format

Unlike most of the other effect devices, Scream 4 features programmable effect presets. Included are a number of factory Patches which can be used as they are or provide you with a good starting point for further tweaking.

Patches use the Windows file extension "*.SM4". Loading and saving Patches is done in the same way as for instrument devices.

Parameters

Scream 4 contains three main sections; Damage (distortion and other types of sound destruction), Cut (EQ) and Body (places the sound in a resonant environment - can serve as anything from a cabinet emulator to a wah-wah to completely new special effects) which can be switched on or off independently. The parameters in each section are as follows:

Common effect device parameters

While the specific parameters for the Scream 4 effect device are described below, some features and procedures are common to all effect devices. Please, refer to "Common effect device features" for information about the Input meter, the Bypass/On/Off switch and Signal Flow Graphs on the effect device.

Damage section controls



The "Damage" section is where you specify the basic sound mangling algorithm and make settings to inflict the desired amount of damage to the sound. There are ten basic algorithms to chose from, ranging from classic distortion effects to digital-sounding warping and modulation effects.

There are five controls in this section, with the following functions:

Parameter	Description
Damage button	This switches the Damage section on or off.
Damage Control knob	This controls the input gain which in turn determines the amount of damage inflicted. The higher the value, the more destruction! When raising the Damage Control you may need to lower the Master level to maintain the same output level (and vice versa).
Damage Type knob	This selects the type of effect - see the table below for a description of the available damage methods.
P1/P2 knobs	The functionality of these knobs vary according to the selected Damage Type - see the table below for a description.

Description of the various Damage Type algorithms

Here follows a basic description of the ten Damage Types available, and what parameters the P1/P2 knobs control for each type:

Type	Description
Overdrive	This produces an analog-type overdrive effect. Overdrive is quite responsive to varying dynamics. Use lower Damage Control settings for more subtle "crunch" effects. - The P1 knob controls the basic tone of the effect. Turn clockwise for a brighter sound. - The P2 knob controls Presence. Presence boosts frequencies in the high midrange *before* the distortion stage which in turn affects the character of the distortion. Turn clockwise for more Presence boost.
Distortion	Similar to Overdrive, but produces denser, thicker distortion. The distortion is also more "even" across the Damage Control range compared to Overdrive. - The P1/P2 knobs control Tone and Presence, respectively - see Overdrive for a description.
Fuzz	Fuzz produces a bright and distorted sound even at low Damage Control settings. - The P1/P2 knobs control Tone and Presence, respectively - see Overdrive for a description.
Tube	This emulates tube distortion. - The P1 knob controls Contour, which is somewhat like a high pass filter, changing the tone and character of the distortion. - The P2 knob controls Bias, which changes the "symmetry" of the tube distortion. Setting this to the minimum or maximum value will produce asymmetrical distortion (typical of a real-life tube amplifier), while a 12 o'clock setting will produce symmetrical distortion (odd harmonics only).
Tape	This emulates the soft clipping distortion produced by magnetic tape saturation and also adds compression which adds "punch" to the sound. - The P1 knob controls Speed, which simulates tape running at different speeds. The higher the Speed setting the more of the original high frequency material in the signal. Turn clockwise for a brighter sound. - The P2 knob controls the amount of Compression. Turning the knob clockwise increases the compression ratio.
Feedback	This effect combines distortion in a feedback loop which can produce many interesting and sometimes unpredictable results. Feedback is basically when a sound source is fed back to itself. An open microphone picking up sound from a nearby loudspeaker that is also being used to amplify sound from the microphone will produce a feedback loop with the associated typical howling. For this effect the Damage Control knob controls the gain of the feedback loop. - The P1 knob controls Size, which could be described as the "length" (i.e. the distance between the microphone and the loudspeaker in the above example) of the feedback loop. - The P2 knob controls Frequency, which for this effect determines which overtones will "howl".
Modulate	Modulate first multiplies the signal with a filtered and compressed version of itself, and then adds distortion. This can produce resonant, ringing distortion effects. - The P1 knob controls Ring, which is the resonance of the filter. Turn clockwise for more pronounced ringing effects. - The P2 knob controls Frequency, which is the filter frequency. Turn clockwise to raise the filter frequency which generally produces a sharper, more piercing effect.



Type	Description
Warp	Warp distorts and multiplies the incoming signal with itself. - The P1 knob controls Sharpness. Lower values will produce a soft, compressed distortion, while higher values produces more harmonics and a sharper sound. - An effect of multiplying a signal with itself is that the fundamental pitch is removed from the signal, leaving overtones only. The P2 knob controls Bias - raise this to reintroduce the fundamental pitch in the sound.
Digital	Lo-fi anyone? This reduces the bit resolution and sample rate for raw and dirty sounds or for emulating vintage digital gear. - The P1 knob controls (bit) Resolution. If the knob is turned fully to the right there is no bit reduction, fully the left the resolution is 1-bit. - The P2 knob controls the sample rate. If the knob is turned fully to the right the sample rate is not reduced; turning it to the left gradually reduces the sample rate.
Scream	Similar to Fuzz, but with a bandpass filter with high resonance and gain settings placed before the distortion stage. - The P1 knob controls the basic tone of the effect. Turn clockwise for a brighter sound. - The P2 knob controls the filter frequency. The high resonance setting of the filter makes it suitable for wah-wah effects.

Cut section (EQ)



The sliders in the Cut section are tone controls, allowing you to cut or boost the level by up to 18dB in the low, mid and high frequency areas. The Cut section is activated with the Cut button above the sliders.

Move the slider from the middle upwards to boost the level, and from the middle downwards to cut the level of the corresponding frequency area.

Body section



The Body section is just what it says - it places the sound in a resonant “body”. Depending on the settings, the result can be similar to a speaker cabinet simulator, an auto-wah effect, or effects with no real-world counterpart. The section is based on 5 basic body types, which simulate how a sound is affected by different physical enclosures. The size and resonance of the Body types can be changed, and the section also features an envelope follower.

The Body parameters are as follows:

Parameter	Description
Body button	This switches the Body section on or off.
Body Type knob	This is used to select one of the five available Body types (A-E).
Body Reso knob	This simulates the resonance of the selected Body. Turning the knob clockwise gives a more resonant effect.
Body Scale	The Body Scale parameter could be said to control the "size" of the Body. Note that this is "inverted" - turning the knob clockwise reduces the emulated size.
Auto knob	Determines the amount of envelope follower effect on the Scale parameter - see below.

About the envelope follower

The Body section features an envelope follower for dynamic control of the Scale parameter. The envelope follower analyzes the amplitude of the incoming signal and changes the Scale setting accordingly - the louder the incoming signal the higher the value of the Scale parameter. The operating frequency (or size) range is set with the Scale parameter, and the envelope follower amount is set with the Auto parameter. A typical use for this is auto-wah effects - try Body Type "B" for a pronounced wah effect.

- **On the back of the Scream 4 you will find an Auto CV output - this delivers the CV signal from the envelope follower, allowing you to dynamically control parameters in other devices. See "Creating a real dynamic wah effect with the envelope follower" for an example.**

About the Master level control



The Master level control should be used when you need to increase or decrease the output level, while retaining the basic character of the effect. It can also be used to balance the level between the distorted sound and the "clean" (unprocessed) sound if the effect is to be switched in and out in the mix.

If the output level is too high, turning down the Damage Control setting would lower the output, but it would also change the character of the distortion, as would changing eq or presence settings.

Simply lowering the mixer channel level (for the channel that Scream 4 is connected to) would also work of course, but this would also mean that the level difference between the unprocessed and processed sound would increase.

So if the clip indicator lights up on the Transport, or if the distorted sound is too loud compared to the normal sound, the solution is to lower the Master output level.

As pointed out elsewhere in the manual, audio out clipping (indicated by the red clip indicator lighting up on the Transport Panel) can only happen in the Reason Hardware Interface. In other words, you never have to worry about levels passed internally from device to device. However, bear in mind that if you use high Master output settings (or a lot of boost in the Cut section) Scream 4 can quite easily cause audio out clipping - and that is most likely not a distortion effect you want!

CV inputs and outputs



On the back of the Scream 4 you will find CV inputs for controlling the following four parameters:

- **Damage Control**
 Use this for dynamically changing the amount of damage effect.
- **P1**
 The use for this depends on the selected Damage Type. For example, if the Feedback effect is selected, this will control the Size parameter - connect it to the CV Out on a Matrix or synth LFO for strange, flanger-like sweeps.
- **P2**
 The use for this depends on the selected Damage Type. For example, if the Scream effect is selected, this will control the Frequency parameter, producing a distorted wah wah sound.
- **Scale**
 Lets you control the Scale parameter in the Body section from another CV source, for wah wah-like effects, etc.

In addition, you find a CV output from the “Auto” (envelope follower) function in the Body section. By connecting this to a CV input for a parameter in another device, the level of the signal going into the Scream 4 will affect that parameter. See below for an example on how to use this.

Screamy tips and tricks

Don't restrict yourself to using Scream 4 as a basic distortion stompbox, but try it in as many ways as possible - you may be surprised to find how often Scream 4 can add power, warmth and color to your sounds. Here are some examples:

Creating a heavy drum sound

Scream 4 is ideal for processing drums. Try connecting it as an insert effect to a Redrum device and experiment with the Damage Types and settings.

- **For classic distorted drums, try the Tube, Tape or Distortion algorithms.**
- **The Scream algorithm is excellent for a really raw body or industrial drum sound.**
- **For more weird, synth-like effects, try the Modulation or Warp effects.**

Remember that you don't have to route the whole drum kit through the Scream device - sometimes it may be better to route the individual outputs from the bass drum, snare and/or toms to a Spider Audio Merger (see "Spider Audio Merger & Splitter"), connect the merged output of the Spider to the Scream 4 and route this to a separate channel in the Mixer. That way, hi-hats, cymbals and similar are unprocessed.

Warming up a mix with the Tape effect

If you find your mixes a bit sterile, the Tape algorithm is excellent for providing some warmth and gentle distortion:

1. **Create a Scream 4 device and connect it between the main outputs of the Mixer and the Audio Hardware device.**
2. **Set the Damage Type to Tape.**
 Make sure the Cut and Body sections are turned off.
3. **Start with a low Damage Control setting and P1 (Speed) and P2 (Compression) at 12 o'clock.**
4. **Play back your mix and adjust the settings.**
 Raise the Damage Control for more tape saturation distortion, adjust P1 to get the desired brightness and raise P2 if you want a more controlled, compressed sound. If you like, you could also activate the Cut section and use the three-band EQ to further adjust the sound.

Using the Body section as a sound enhancer/phaser/wah

Nothing stops you from using the Body section on its own, without Damage. Try this:

1. **Create a sampler device (e.g. an NN-19) and select an electric piano patch.**
2. **Select the sampler and create a Scream 4.**
 It is added as an insert effect.
3. **Turn off the Damage section and instead activate the Body section.**

You will find that this adds a resonant character to the sound, which will make it more "alive" and help it stand out in a mix. You should experiment with the Body settings to find the character that suits you best. You could also activate the Cut section- if you e.g. find the sound too bassy, just lower the "Lo" slider a bit.

4. **Now flip the rack around and connect the CV out from the sampler's LFO to the Scale CV input on the Scream 4.**
 As you can hear, the Scale is modulated by the LFO.
5. **Adjust the amount of Scale modulation with the pot next to the CV input on the back of the Scream 4, and the speed (and waveform) of the modulation in the LFO section on the sampler device.**
 With this type of modulation setup, it's easy to get lush, dreamy phaser effects. For a more wah wah-like sound, select Body type B and raise the Reso and Scale settings.



Emulating vintage digital gear

The first generations of digital instruments (drum machines, synths and samplers) used 8 or 12 bit sampling and processing, at low sample rates. This "lo-fi" sound is still in high demand, e.g. in hip-hop and R'n'B. Try this:

1. **Connect a Scream 4 as an insert for a Redrum, with a suitable kit selected.**
2. **Set Damage Type to Digital and turn P1 and P2 fully right.**
3. **Play back and lower P1 (bit depth) and P2 (sample rate) to get the sound you want.**
 You may also want to use the Cut section to emphasize or cut frequencies in the sound.

Creating a real dynamic wah effect with the envelope follower

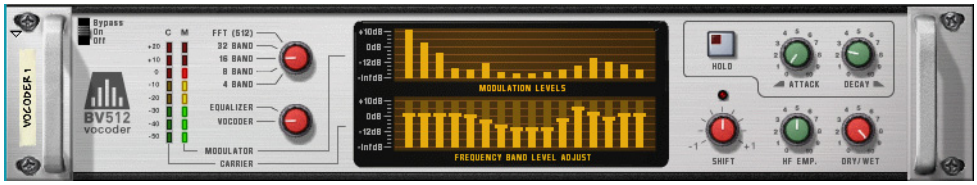
As we have shown, you can get auto-wah-like effects with the Body section in Scream 4 (by using the Auto parameter). You could also use the ECF-42 envelope controlled filter and trigger this with a gate signal - this is after all a "real" filter and sounds even more like a wah effect. However, to get a "real" auto-wah effect that responds to the signal level, you need to combine both these devices:

1. **Create an instrument device that you want to process with an auto-wah.**
 It should be velocity responsive so that the harder you play, the louder it sounds.
2. **Create a Scream 4 device and an ECF-42 device.**
 Both these should now be connected as insert effects to the instrument device.
3. **Turn off all three sections in the Scream 4.**
 This is of course a matter of taste - but here we will show how to use the envelope follower in Scream 4, not its sound destruction capabilities.
4. **Flip the rack around and connect the Auto CV Output on the Scream to the Freq CV input on the ECF-42.**
5. **Lower the pot next to the CV input a bit - the envelope follower is rather sensitive and you probably don't want the filter to open too much.**
 You can adjust this later if needed.
6. **On the ECF-42, select the BP 12 (bandpass) mode and set the Res setting rather high.**
7. **Play the instrument device and adjust the Freq setting on the ECF-42 to taste.**
 As you can hear, the harder (or the more notes) you play, the more the filter will open.

- **If you find the auto-wah too responsive, you could add a compressor between the instrument device and the Scream 4 to even out the level differences a bit.**
- **The Spider CV Merger and Splitter (see "Spider CV Merger & Splitter") can be used to invert and split the Auto CV output for even greater flexibility.**

Chapter 35
BV512 Vocoder

Introduction



The BV512 is an advanced vocoder device with a variable number of filter bands. It also has a unique 1024-point FFT vocoding mode (equivalent of 512-band vocoding) for very precise and high quality vocoded speech. By connecting the BV512 to two instrument devices, you can produce anything from vocoded speech, singing or drums to weird special effects.

Even if you have worked with a vocoder before, please read the following section. Knowing the basic terms and processes will make it much easier to get started with the BV512!

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

How does a vocoder work?

Carrier and modulator

A vocoder accepts two different input signals, a "carrier" and a "modulator". It analyzes the modulator signal, applies its frequency characteristics to the carrier signal and outputs the resulting "modulated" carrier signal.

In the most typical case, the carrier signal is a string or pad sound and the modulator signal is speech or vocals - the result will be a talking or singing synth sound. The modulator could also be drums or percussion (for rhythmically modulated sounds and effects) or any sound with changing frequency content.

Filter bands

Technically, a vocoder works in the following way: The modulator signal is divided into a number of frequency bands by means of bandpass filters (called the "modulator filters" or "analyzing filters"). The signal in each of these bands is sent to a separate envelope follower (which continuously analyzes the level of the signal). The carrier signal is sent through the same number of bandpass filters (the "carrier filters"), with the same frequency ranges as the filters for the modulator signal. The gain of each bandpass filter is controlled by the level from the corresponding envelope follower, and the filtered signals are combined and sent to the vocoder's output.

In this way, the carrier is filtered to have roughly the same frequency characteristics as the modulator. If the modulator signal has a lot of energy in one of the frequency bands, the gain of the corresponding filter band for the carrier signal will be high as well, emphasizing those frequencies in the output signal. If there is no signal at all within a frequency band in the modulator signal, the corresponding band in the output signal will be silent (as the gain will be zero for that filter).

There are several factors determining the quality of the vocoder sound, but the most important is the number of filter bands. The larger the number of filter bands, the closer will the output signal follow the modulator's frequency characteristics. The BV512 offers 4, 8, 16 or 32-band vocoding.

► **Even if a high number of bands will make the sound more precise and intelligible, this isn't always what's desired! Vocoding with a lower number of bands can give results that sound different, fit better in a musical context, etc.**

FFT vocoding

The BV512 has an additional FFT mode, in which the vocoding process isn't based on bandpass filters as described above. Instead, FFT (Fast Fourier Transform) analysis and processing is used. This equals 512 "conventional" frequency bands and results in a very precise and detailed vocoder sound. Note:

- **The FFT mode is best suited for vocoding speech or vocals, giving crystal clear and highly intelligible results.** It is not so well suited for vocoding drums and percussion, since the FFT process is inherently "slower" than the regular filtering and doesn't respond as quickly to transients, and also there will be a slight delay added to the signal (in the region of 20ms). A workaround solution to this would be to move the modulator signal slightly ahead to compensate for the delay.
- **Where the conventional filter bands are distributed logarithmically (i.e. the same number of filter bands per octave), the 512 bands in the FFT mode are distributed linearly.** This means a lot of the bands will be in the high frequency range - this is one of the reasons for the clear sound but it is also something to keep in mind when making settings for the vocoder in FFT mode.

Setting up for vocoding

! **Note that the vocoder set-up might differ, depending on which DAW you are using. The important thing in the vocoder set-up is to be able to route MIDI to an instrument to play the carrier sound - and at the same time also be able to route the modulator audio (e.g. your voice) to the vocoder. This might require both an instrument track and an audio track in your DAW. Consult the manual for your DAW for details.**

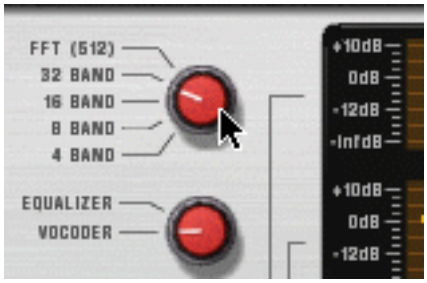
Here are some guidelines for creating a typical vocoder setup. We assume here that you have a MIDI keyboard connected. For details on the parameters, see "BV512 parameters".

1. **Create the instrument device you want to use for the carrier signal.** This could typically be a synth or a sampler. In this example we choose a Subtractor synthesizer.
2. **Set up the carrier device (Subtractor) for a sustaining, bright sound.** It's important to have high frequencies (a lot of harmonics) in the carrier. On the Subtractor, a simple but effective carrier sound would be based on a sawtooth wave, with the filter fairly open. For more about choosing carrier sounds, see "Choosing a carrier sound".
3. **Select the carrier device (Subtractor) and create a BV512 Vocoder.** If you flip the rack around you will see that the carrier input of the Vocoder is automatically connected to the Subtractor output:

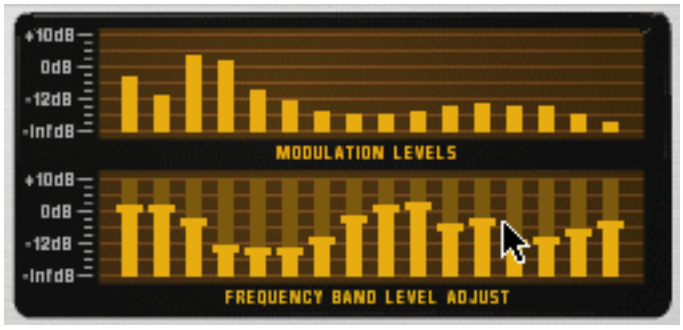


4. **Route the microphone/audio signal to the Modulator Input on the BV512.** Depending on which DAW you are using this routing could be a little different. If possible, connect the appropriate Audio In on the I/O device to the Modulator input on the back of the BV512.
5. **Make sure the "Dry/Wet" knob on the BV512 Vocoder is turned to "Wet" (fully clockwise).**
6. **Play some notes or chords on your MIDI keyboard and sing through the microphone.** The result should be the classic vocoded vocal sound.

7. **Try the different filter band options and note the difference in sound.**



8. **You can also adjust the vocoder sound by clicking and dragging the bars in the lower display.**
 Each bar corresponds to a frequency band, with low frequencies to the left and high frequencies to the right. You adjust the level of a band by dragging its bar up or down. Clicking and dragging across the bars allow you to change the levels of several bars, much like drawing an eq curve.



The upper display shows the spectrum of the modulator signal, for display only.

→ **To reset a band to ±0 dB, press [Ctrl](Win) or [Cmd](Mac) and click on it.**
You can also reset all bands to zero by bringing up the context menu for the Vocoder device and selecting "Reset Band Levels".

9. **If the vocoder sound is "muddy" or indistinct, try raising the "HF Emph" knob on the Vocoder.**
 This parameter (High Frequency Emphasis) boosts the high frequencies in the carrier signal.

10. **Try out the other parameters if you like.**
 See "BV512 parameters" for details.

Using the BV512 as an equalizer

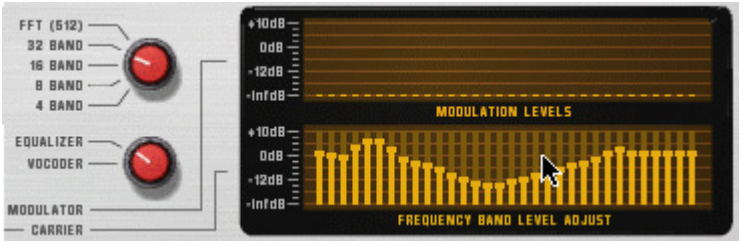
The BV512 has a unique equalizer mode, in which the device works purely as an insert effect (the modulator input isn't used). This allows you to use the processing filters of the vocoder as a kind of graphic equalizer.

Setting up

1. **Select the device that you want to process through the BV512.**
2. **Create a BV512 device.**
 It is automatically connected to the instrument output(s), using the Carrier Input jacks.
3. **Set the switch to the left of the displays to "Equalizer".**



In use



In equalizer mode, you cut or boost frequencies by clicking and dragging in the lower display - just as with a regular graphic equalizer. The usage and results differ depending on which mode is selected:

- **4 - 32 band mode**
 As in vocoder mode, the number of bars in the display conforms to the number of bands selected (4, 8, 16 or 32). With a higher number of bands you get a more detailed control over the frequency response. However:

→ **In these modes, the equalizer will "color" the sound even if all bands are set to ±0 dB!**
This is due to phase interaction and overlap between the bandpass filters.

Therefore you probably want to use the 4 - 32 band mode for coloring and mutating sounds - not for subtle, "clean" equalizing.

- **FFT (512) mode**
 In FFT (512) mode you still get 32 bars in the display, but the each bar may control several frequency bands (remember that there are 512 bands in FFT mode). Since the frequency bands are distributed linearly in FFT mode, bars to the left in the display control few frequency bands while bars to the right control many frequency bands.

→ **In FFT (512) mode, setting all bands to ±0 dB is the same as bypassing the equalizer - the sound will not be affected.**
This makes FFT mode suitable for "clean" equalizing, where you want to boost or cut some frequencies without changing the basic sound character.

→ **However, FFT mode equalizing is not suited for very drastic frequency cuts or boosts, as this may give audio artefacts due to the workings of FFT processing.**
Still: as always, there are no hard and fast rules. Let your ears judge!

→ **Keep in mind that FFT mode also introduces a slight delay to the signal.**

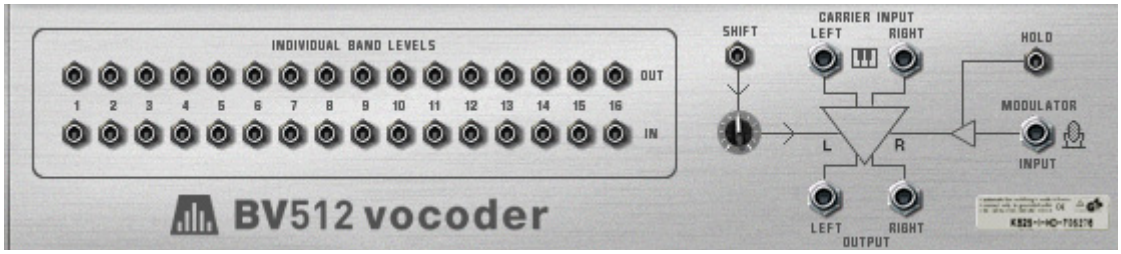
BV512 parameters

On the front panel of the BV512 Vocoder, you will find the following parameters and displays:

Parameter	Description
Bypass/On/Off switch	In Bypass mode, the carrier signal passes through the device unaffected and the modulator signal is disregarded. In On mode, the device outputs the vocoded or equalized signal. Off mode cuts the output, silencing the device.
Level meters	Show the signal level of the carrier and modulator signals, respectively.
Band switch	Selects the number of filter bands (4, 8, 16 or 32) or FFT (512) mode.
Equalizer/Vocoder switch	Determines whether the BV512 should work as a vocoder or an equalizer. In Equalizer mode, the Modulator input is disregarded.
Modulation level display	The upper display shows the spectrum of the modulator signal.

Parameter	Description
Frequency band level adjust	The lower display allows you to adjust the level of each filter frequency band, by clicking and dragging the corresponding bar. In vocoder mode this affects the vocoded sound. In equalizer mode, this is where you cut or boost frequencies. To reset a band to ±0 dB, press [Command] (Mac) or [Ctrl] (Win) and click on its bar in the display. To reset all bands, select "Reset Band Levels" from the device context menu. Note: when FFT (512) mode is selected, each of the 32 bars in the display corresponds to several frequency bands, with bars to the right in the display controlling progressively more bands (due to the FFT bands being linearly distributed over the frequency range).
Hold button	Clicking this button "freezes" the current filter settings. While the button is lit, the modulator signal doesn't affect the sound - the carrier signal is filtered with the settings as they were the moment you activated Hold. Click the button again to turn off Hold. Hold is also automatically reset (turned off) when you stop sequencer playback - just like the pitch bend and modulation wheels on synth devices. This function can be controlled via CV or MIDI, for sample and hold-like effects. The Hold button is not available in Equalizer mode.
Attack	This is a global attack time control, affecting all envelope followers (see "Filter bands"). Normally you probably want this set to zero, to make the vocoder react as quick as possible. Raising the Attack time can be useful for "smearing" sounds, creating pads, etc. Not available in Equalizer mode.
Decay	Similarly, this controls the decay time for all envelope followers, i.e. how quick the filter band levels drop. Adjust this according to taste and context. Not available in Equalizer mode.
Shift	Shifts the carrier filters up or down in frequency, drastically changing the character of the vocoded (or equalized) sound. This parameter can be controlled via CV, for phaser-like sweeps and special effects.
HF Emph (High Frequency Emphasis)	Boosts the high frequencies in the carrier signal. This is sometimes desired to get a clearer vocoded sound. The reason is that a carrier signal should theoretically contain roughly equal energies in all frequency ranges for best results - in a typical synth sound the high frequencies are often weaker than the low frequencies. Raising the HF Emph control will rectify this. Not available in Equalizer mode.
Dry/Wet	Determines the balance between modulator sound (dry) and vocoded sound (wet). To get the pure vocoder sound, set this to wet (turned fully right). Not available in Equalizer mode.

Connections



The back panel of the BV512 offers the following connections:

Individual band levels

These are CV outputs and inputs.

- **The upper row outputs CV signals generated by the envelope followers for each frequency band.**
- **The lower row are CV level inputs to the individual bandpass filters through which the signal is processed (the "vocoder filters").**
 Connecting a CV signal to one of the inputs breaks the internal signal path from the corresponding envelope follower (in other words, that frequency band is now controlled by the CV signal you've connected - not by the corresponding frequency band in the modulator signal).
- **If 16 band mode is selected, each output/input pair corresponds to a separate frequency band.**
 In 8 band or 4 band mode, only the 8 first or 4 first output/input pairs are used. In 32 band mode, each output is a mix of two adjacent frequency bands and each input controls two bands. Finally, in FFT (512) mode each output/input pair corresponds to several frequency bands.

There are several interesting uses for the Individual band levels connectors: you can cross-patch frequency bands so that e.g. low frequencies in the modulator signal controls high frequency bands in the vocoder, you can extract CV signals for controlling synth parameters in other devices, you can base the vocoding on CV signals from other devices rather than on a modulator signal, etc. See "Using the individual band level connections" for details.

Other CV connections

Connection	Description
Shift (CV in)	This allows you to control the Shift parameter from an external CV source. A sensitivity knob determines how much the Shift setting is affected by the CV signal.
Hold (Gate in)	When a gate signal is sent to this input, the Hold function is activated (see "Hold button"). Hold remains on until the gate signal "goes low" (falls to zero). By connecting e.g. a Matrix to this input, you can create "stepped" vocoder sounds, sample and hold-like effects, etc.

Audio connections

Connection	Description
Carrier input	This is where you connect the instrument device that provides the carrier signal (or the device to be processed in Equalizer mode) - typically a synth or sampler device. The vocoder can handle mono or stereo carrier signals.
Modulator input	This is where you connect the instrument device that provides the modulator signal, in mono. This connection is not used in Equalizer mode.
Output	In Vocoder mode, the outputs carry a mix between the vocoded signal and the modulator signal (as set with the Dry/Wet control on the front panel). In Equalizer mode the output is the carrier signal, processed through the equalizer filter. Note that the output will be in mono if the Carrier input is in mono, and vice versa - the BV512 does not process mono into stereo.

Tips and tricks

Choosing a carrier sound

As always, which carrier sound to choose is a matter of taste and musical context. However, here are a few guidelines to help you get a good result:

- **The carrier sound should preferably have a lot of harmonic content (brightness) - dark or muffled sounds will not "give the vocoder much to work with".**
- **Often, you want the carrier sound to sustain at an even level (i.e. it shouldn't "die out" when you hold a chord). Similarly, you most often want a reasonably fast attack (although not with a distinct, sharp click or edge).**
- **You may want a sound that is rather static over time, without drastic envelope control of filter cutoff for example.**
- **If you want to play vocoded chords, the carrier sound must of course be polyphonic.**

Here are some hands-on suggestions for carrier sounds:

→ **A simple Subtractor pad based on a sawtooth waveform.**
You could simply start with the initial patch (as set up when you create a new Subtractor device). Open the filter, turn off envelope modulation of the cutoff frequency and raise the Amp Envelope Sustain.
If you want a classic, rich chorus-like sound, use two detuned oscillators - or better still, add a UN-16 Unison device as an insert effect between the Subtractor and the vocoder!



A simple but effective carrier sound setup.

→ **A similar fat carrier sound can be obtained using a Malström device with a patch based on the "Sawtooth*16" graintable.**
With the Malström you can get a stereo carrier signal with no extra devices: simply select the "Sawtooth*16" graintable for both oscillators, detune the oscillators slightly with the Cent controls and raise the Spread parameter to the desired stereo width. No filter routings are necessary.

→ **For a more distinct and precise sound, try using a narrow pulse waveform.**
You get this by selecting e.g. a sawtooth wave on the Subtractor, setting the Phase Mode selector to "–" and turning the Phase knob to the left until you get the desired sound. This type of carrier sound lends itself well to monophonic vocoder lines in the lower registers.

→ **Use noise as a carrier.**
Try using pure noise (possibly filtered down a bit) for robotic voices, whispering and special effects. It's also very useful to add a bit of noise to a sawtooth or pulse sound - this makes vocoded speech clearer and more intelligible.

→ **Use sampled strings or choir sounds.**
A rich drawbar organ sample can also be a cool carrier sound.

→ **For unusual vocoder sounds, try using the Malström as carrier device, e.g. with a glassy, digital pad sound selected.**
Try turning up the Attack and Decay controls on the BV512, for smeared, rhythmic or pseudo-random modulation of a pad.

Choosing a modulator sound

The modulator sound should typically have varying level and harmonic content. As we've already mentioned, the most typical modulator sounds are vocals or speech and drums or percussion.

→ **The quickest way to get an "internal" modulator sound is to use a rhythmic loop in the Dr. Octo Rex device.**
This way you don't have to program a rhythm pattern. On the other hand, using a Redrum as modulator allows you to create exactly the rhythm you want and fine-tune the sounds and the groove.

Using the modulator as carrier

You can get cool special effects by using the same device both as carrier and as modulator. For example, try processing a Redrum device in the following way:

1. **Create a Redrum device and set up the desired patch and pattern.**
2. **Create a Spider Audio Merger & Splitter device.**
3. **Create a BV512 Vocoder.**
4. **Flip the rack around and connect the devices in the following way:**



The output of the Redrum goes into the splitter section of the Spider, and is split into two signals. One signal goes into the carrier input of the vocoder, the other goes into the modulator input.

This is essentially the required connections, but for best results it's a good idea to add some distortion and/or compression to the carrier signal - this increases the amount of high frequencies in the carrier signal:

5. **Press [Shift] and create a Scream 4 distortion device.**
6. **Connect the distortion device as an insert effect between the Spider and the carrier input of the vocoder.**
 Now, the carrier signal will be processed in the distortion device, but not the modulator signal.
7. **Play back the pattern and experiment with the settings on the vocoder and distortion device.**

→ **This technique can also be used to process vocals and speech.**

→ **Try adjusting the Shift parameter for new effects and sounds.**
Remember that you can route CV to the Shift parameter on the back of the BV512 - use e.g. a Matrix or an LFO output on a synth device!

Controlling the Hold function

As described in "Hold button", pressing the Hold button on the front panel "freezes" the current filter spectrum until you deactivate it again. This can be used for creating sample & hold-like effects, stuttering or garbled vocoder sounds:

→ **Connect e.g. the Gate output on a Matrix device to the Hold input on the back of the BV512.**
By playing back a gate pattern on the Matrix, the Hold function will repeatedly be turned on and off according to the programmed rhythm in the pattern. Hold will be active for the length of each gate signal.

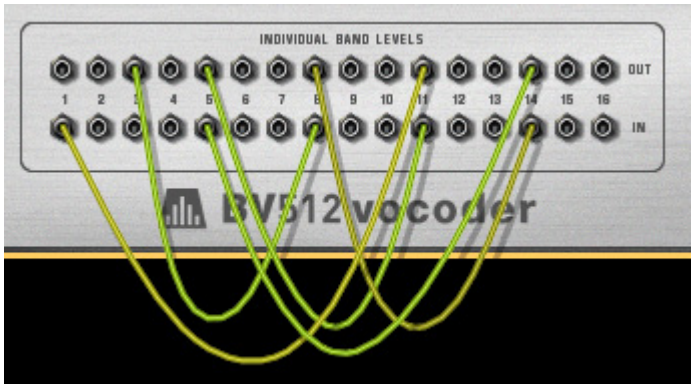
→ **Automate the Hold function with the main sequencer, either by recording it or by drawing in its controller lane.**

- **If you route MIDI to the BV512 you can control the Hold function in two ways by default: By pressing a damper pedal connected to your MIDI controller or by playing the note C4.**
 In both cases, the Hold function will be momentary - Hold is on until you release the pedal or key.

Using the individual band level connections

As described in "Individual band levels", the individual band level connectors on the back are CV output and input jacks. The upper row sends out the CV signals from the envelope followers for the different frequency bands, while the lower jacks are CV inputs for controlling the individual bandpass filters (breaking the internal connection from the envelope followers). There are several interesting things you can do with these connections:

Crosspatching frequency bands



By connecting outputs to inputs in alternative configurations, you can drastically change the result of the vocoding. For example, you could have low frequencies in the modulator signal give high frequencies in the vocoded sound and vice versa. Note:

- **In 4 band and 8 band mode, only the 4/8 first output/input pairs are used.**
- **In 32 band mode and FFT (512) mode, each connection corresponds to two or several frequency bands.** This means that connecting an output to the input with the same number is not the same as using the internal signal path (no CV cable connected). You can hear this quite clearly in FFT (512) mode: connect all outputs to the corresponding inputs and gradually remove the CV cables while listening to the vocoder sound - the sound will progressively get more detailed.

Extracting CV from the vocoder

You can connect an individual band level output to any CV input on any device. This means you can use the vocoder as an envelope follower, having elements in the modulator sound control a parameter in another device, e.g. an effect. Note:

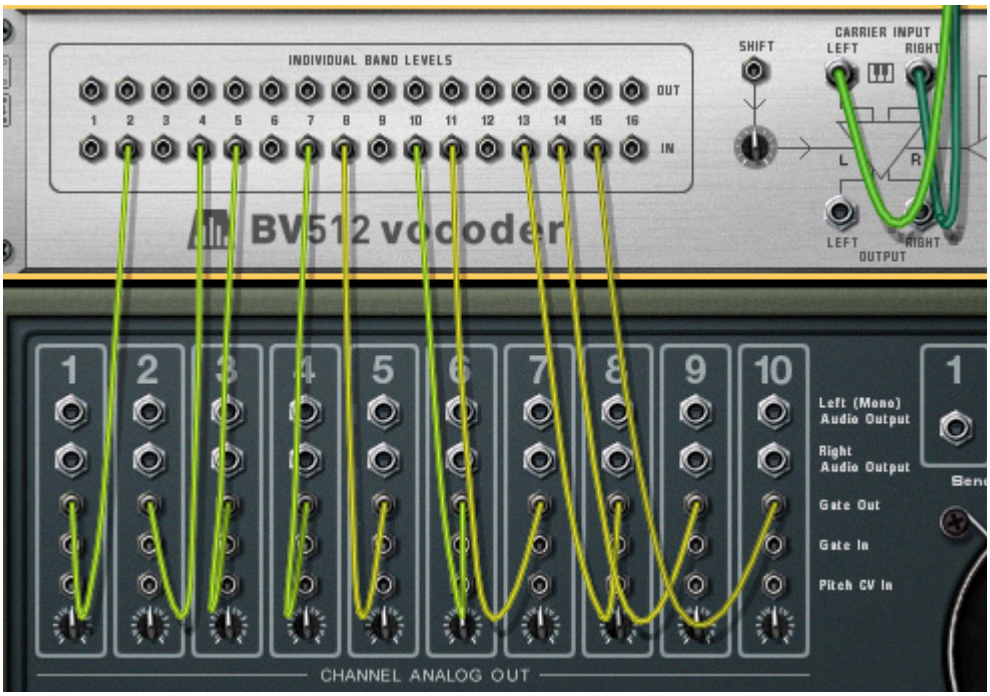
- **The Attack and Decay settings on the BV512 panel affect the envelope followers, and thus the rise and fall times of the CV signals from the individual band level outputs.**
- **If you are using the vocoder in a mode with many bands, but want a broader frequency range to generate the CV signal, you can merge several band outputs into one CV signal - use a Spider CV Merger & Splitter device.**

Controlling vocoder bands from an external source

Connecting a CV source to an individual band input breaks the internal connection from the corresponding envelope follower. This way you can “manually” control the vocoder filters. Some applications:

→ **Connect the CV outputs for one or more envelopes in the carrier device to individual band inputs.** When you play the carrier instrument, one or more of the bandpass filters in the vocoder will automatically open, adding an extra attack to the sound. Useful if you really want to “play” the carrier, rather than just hold a chord.

→ **Connect the gate outputs on a Redrum to individual band level inputs.**
With this connection (and no device connected to the Modulator input), the Redrum will serve as a pattern sequencer, opening and closing different filter bands. To adjust the gate times, set the drum sounds to Gate mode and use the Length parameter. The result is totally different from using the audio signal of the Redrum as modulator.



The vocoder bands are now solely controlled by the gate signals from the drum channels - the modulator input isn't used. Note that you can use a Spider CV Merger & Splitter device to split a gate signal, sending it to several bands. Also, note that the velocity of the programmed drum notes govern the level of the corresponding filter bands.

"Playing" the vocoder from a MIDI keyboard

! **To route MIDI notes to the BV512 in Reason Rack Plugin, you need to put it in a Combinator and turn on "Receive Notes" in the programmer (see "About the Key (receive MIDI Notes) column" in the Combinator chapter).**

If you have routed MIDI to the BV512, playing notes from C1 and up will control individual filter bands. For example, in 16 band mode, C1 controls band 1, C#1 band 2 and so on up to D#2 (which controls band 16).

- **The level of the bands is proportional to key velocity (how hard you play).**
- **A band will be "open" until you release the corresponding key.**
- **Bands to which you have connected a CV signal (using the individual band level inputs on the back panel) will not respond to MIDI keys.**

Note that with this function, you "play the modulator bands". However, you still need both a carrier and a modulator signal to get any sound. Typically, you would first record the notes or chords for the carrier device in the sequencer, then create a sequencer track for the vocoder and "play" it from your MIDI keyboard while playing back the recorded carrier notes and at the same time inputting a signal on the Modulator input.

← **An interesting application of this is to patch the vocoder as an insert effect for the whole mix (the output of the main mixer connected to the carrier input), and "play the vocoder" while inputting a signal on the Modulation input. Only the frequency bands for which you press keys will be attenuated. Use the FFT (512) mode for best results.**

Using the BV512 as a reverb

This is a very special trick which can be quite cool. Proceed as follows:

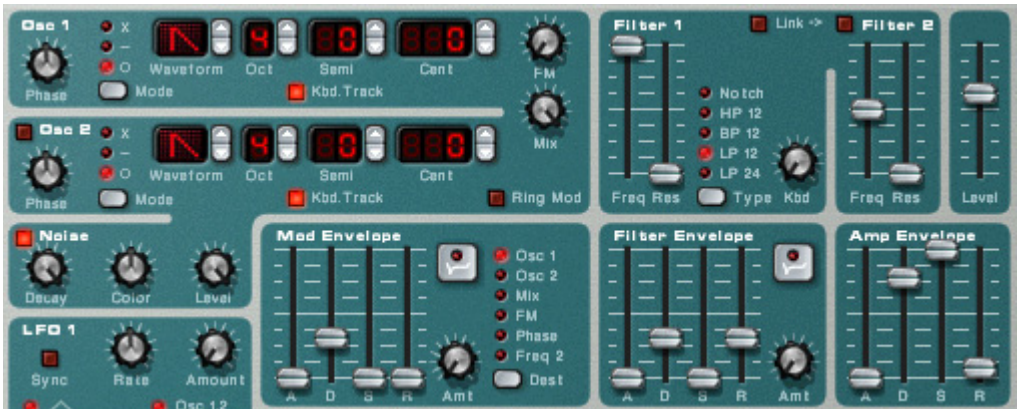
1. **Create a Mixer 14:2 device from the Utilities Palette.**
2. **Create a Redrum device.**
 The “vocoder-reverb” is best suited for drums, even though nothing stops you from using it on other sounds.
3. **Create a Subtractor and a vocoder.**
 The Subtractor will automatically be routed to the carrier input. We don’t need a dedicated modulator device in this setup.
4. **Flip the rack around and connect Aux send 1 on the Mixer to the modulator input on the vocoder.**
5. **While you’re there, re-route the vocoder output to Aux return 1.**
 This way, our vocoder-reverb will be connected as a regular send effect.



6. **Set the vocoder to FFT (512) mode, turn the Decay knob to between 6 and 7 and turn the Dry/Wet control to “wet” (fully right).**
7. **On the Subtractor, set up a noise sound as follows:**

- **Turn the Oscillator Mix knob fully to the right.**
- **Turn on the Noise section (but make sure Osc 2 is off).**
- **In the Noise section, turn Color to around twelve o’clock.**
- **Open the filter fully and make sure resonance is set to 0.**
- **Make sure Filter Envelope Amt is 0 (and turn off velocity modulation).**

→ **Raise the Sustain to full in the Amp Envelope section.**



Now we want the Subtractor to play a continuous noise. You could just route MIDI to it, play a note and keep it pressed, but that will probably wear you out in the long run. Better to use a Matrix:

8. **Create a Matrix and route it to the Subtractor.**
 We really only need the Gate connection - the note number isn't important with the noise patch.
9. **Set up a one step pattern with a tied gate (press [Shift] and draw the gate) and start playback on the Matrix.**
 Now the vocoder gets a continuous noise signal as carrier.
10. **Create a suitable drum pattern on the Redrum and start pattern playback.**
11. **Gradually turn up send 1 for the Redrum channel in the mixer.**
 This now serves as a balance control between the dry drum sound and the reverb, generated by the voccoded noise! Set it to a suitable reverb level.
12. **Use the Decay control on the vocoder to adjust the reverb decay time.**
13. **Use the Noise Color control on the Subtractor to make the reverb darker or brighter.**
 You could use the filter cutoff for this as well.

That's it - a pretty good reverb sound with a lot of control. Although the settings above give the most natural sound, you can vary the sound and create special-FX reverb in the following ways for example:

- **Switch the vocoder to a lower band mode.**
- **Lower the cutoff and add some resonance in the Subtractor filter.**
- **Modulate the Subtractor filter with a fast LFO.**
- **Set the Subtractor filter to HighPass mode to remove the bottom end from the reverb.**
- **Turn off the Matrix controlling the Subtractor and "play" the noise patch yourself (or from the sequencer). This way you can create gated reverb effects, etc.**

Creating a stereo reverb

What you've got above is a mono reverb. Here's how to make it stereo:

1. **Select the Subtractor and create a Spider Audio Merger & Splitter device.**
2. **Create a DDL-1 delay.**
3. **Connect the devices in the following way:**
 The Subtractor output should be routed to a Splitter input on the Spider. One split output should be routed to one of the carrier inputs on the vocoder, the other split output should be routed to the delay. The delay output (mono) should be routed to the other carrier input on the vocoder.



The vocoder will now get a "fake-stereo" noise carrier signal.

4. **Make sure the output from the vocoder is connected in stereo to the Aux return on the Mixer.**
5. **Finally turn down the Feedback on the delay, set it to all "wet" and set the decay time to a second or so.**

When you now start playback on the Redrum, the reverb will be in stereo!

Chapter 36
RV7000 Mk II
Advanced Reverb

Overview



The RV7000 Mk II is a high quality reverb processor. It features ten different reverb and echo algorithms, ranging from rooms and halls to special effects. The Mk II version also incorporates a high-quality zero-latency convolution algorithm, which makes it possible to load sampled impulse responses - and even sample and use your own impulse responses!

Since the RV7000 Mk II comes with a number of useful reverb presets, you could simply select one and tweak the most important parameters on the main panel - or you could use the Remote Programmer panel to fine-tune the reverb in great detail.

The RV7000 Mk II also contains an equalizer and a gate section. Both of these are for processing the actual reverb sound, making it possible to get virtually any kind of reverb character, including gated reverb.

About the Patch format

The RV7000 Mk II features programmable effect presets. In the Factory Sound Bank you will find a number of preset Patches which can be used as they are or provide you with a good starting point for further tweaking.

Patches use the Windows file extension "*.RV7". Loading and saving Patches is done in the same way as for instrument devices.

! **Songs and/or Patches that use Convolution (or have an Impulse Response sample loaded) can not be opened in older Reason versions. The same goes for Combi Effect patches that contain RV7000 Mk II in Convolution mode (or with a sample loaded, or with the Combi programmer referencing Convolution parameters).**

! **Convolution patches that use samples only contain references to the samples - not the sample itself. This is the way all sampler devices in Reason work.**

Connections

Typically you connect the RV7000 Mk II as a send effect, as this allows you to use it for processing several different mixer channels. However, it's also possible to use it as an insert effect - use the Dry/Wet control on the main panel to adjust the balance between the dry, unprocessed sound and the reverb. Note:

→ **The RV7000 Mk II is a true stereo reverb (except in Convolution mode), which means that it will use the stereo input information when processing both channels (without summing the input channels).**
It's also possible to use it as a mono in - stereo out effect. Which type of connection to use (mono or stereo in) depends on the material. If the audio sources are in mono (or in stereo but with no important difference between the left and right channel) using a mono input is sufficient.

→ **If you want to use RV7000 Mk II's Reverse reverb effect, you should consider connecting it as an insert effect or using a Send on the Mixer, with Pre-fader mode selected (and the channel fader lowered).**
This is because you typically don't want to hear the dry sound when using the Reverse effect. See "The Gate section".

The main panel

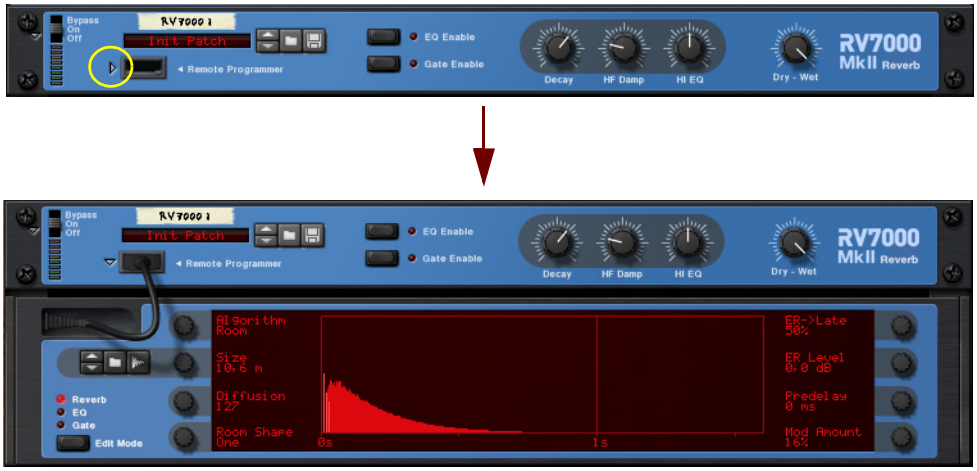


The RV7000 Mk II main panel.

When you create an RV7000 Mk II, only the main panel will be shown. This contains a section for handling patches, on/off buttons for the EQ and Gate sections, the most important reverb parameters and a dry/wet mix control. To select a reverb patch and make coarse adjustments, this is all you need.

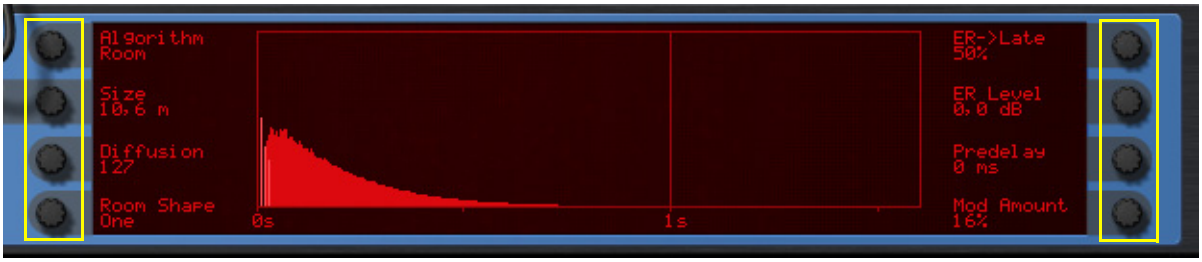
The Remote Programmer

Clicking the arrow button next to the “cable slot” on the main panel brings up the remote programmer panel.



This is where you make detailed settings for the reverb. Note:

- **The Edit Mode button to the left determines which section to make settings for, Reverb, EQ or Gate.**
- **Settings are made with the eight dials around the graphic display. The functions of the dials differ depending on the selected Edit Mode and the selected reverb algorithm. Next to each dial, the display shows the name and value of the corresponding parameter.**



- **Not all modes and algorithms use all eight dials. If a dial isn’t used in the selected mode, nothing will be shown next to it in the display.**
- **You cannot make settings in the graphic display itself - this is for showing a graphic representation of the selected reverb.**

Reverb algorithms and parameters

Common effect device parameters

While the specific parameters for the RV7000 Mk II effect device are described below, some features and procedures are common to all effect devices. Please, refer to "Common effect device features" for information about the Input meter, the Bypass/On/Off switch and Signal Flow Graphs on the effect device.

About the main panel parameters



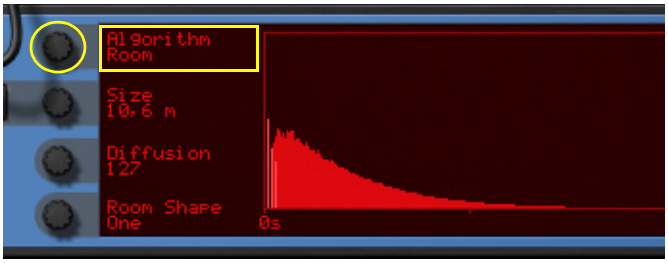
On the main panel you find three parameters that are available for all algorithms:

Parameter	Description
Decay	This governs the length of the reverb or the feedback if an echo algorithm is selected.
HF Damp	Controls how quickly the high frequencies should decay in the reverb. Raise it to gradually remove high frequencies, making the reverb sound warmer and less bright.
HI EQ	This is a high-shelving EQ that works much like a typical treble control on a mixer or amplifier. Lower the setting for a softer reverb sound or raise it to get more high frequencies.

Selecting an algorithm

You select a reverb algorithm in the remote programmer panel:

1. **Click the remote programmer arrow button on the main panel to display the remote programmer panel.**
2. **Make sure the Edit Mode button is set to Reverb.**
3. **Use the top left dial to select a reverb algorithm.**
 The selected algorithm is shown in the display next to the dial.



Here's a quick overview of the ten algorithms - for details and parameter descriptions, see below.

Algorithm	Description
Small Space	Emulates a small enclosed space (a small room or a resonant body).
Room	Emulates a room with adjustable shape and wall character.
Hall	Emulates a hall.
Arena	Emulates a large arena, with separate pre-delay for the left, right and center reverbs.
Plate	Emulates a classic plate reverb.
Spring	Emulates a spring reverb, as used in e.g. guitar amplifiers.
Echo	An echo effect with gradually diffusing echo repeats. Can be synced to Reason's tempo.
Multi Tap	A multi-tap delay with four different delay lines and tempo sync.
Reverse	A reverse reverb that "pushes" the dry sound to appear after the reverb. The result is a backwards reverb leading up to the direct sound.
Convolution	The zero-latency Convolution algorithm uses impulse response samples. The samples are used for generating the desired reverb effect (or actually any type of effect - depending on what sample you use).

Small Space

This algorithm places the sound in a small enclosed space, ranging from a tiny resonant body to a room. The parameters are:

Parameter	Description
Size	The size of the emulated space.
Mod Rate	The reverb can be randomly modulated for a more even sound (or for special effects). This parameter sets the rate of modulation (the amount is set with Mod Amount).
Room Shape	Select from four different room shapes, affecting the character of the reverb.
LF Damp	Controls how quickly the low frequencies should decay in the reverb. Raise it to gradually remove low frequencies, making the reverb sound "thinner" and less boomy.
Wall Irreg	Adjusts the positioning of the emulated walls in the small space. The lowest setting emulates two directly opposed walls while higher settings emulate more walls and angles, for a more complex resonance.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the reverb.
Mod Amount	Sets how much the reverb will be modulated. Use fairly low settings when emulating real rooms and resonant bodies, and higher settings for special effects.

Room

Emulates a medium-sized room, with the following parameters:

Parameter	Description
Size	The size of the emulated room.
Diffusion	At low Diffusion settings, you will hear the individual reverb "bounces" more clearly, while higher settings produce a more "smeared", dense and even reverb.
Room Shape	Select from four different room shapes, affecting the character of the reverb.
ER->Late	The first "answers" in the reverb are called early reflections (ER) and are typically more pronounced than the actual reverb tail. This parameter sets the time between the early reflections and the reverb tail. This is set as a percentage - the actual delay time depends on the Size setting.
ER Level	Adjusts the level of the early reflections. "0" is normal level.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the early reflections and reverb.
Mod Amount	Sets how much the reverb will be modulated. Moderate modulation gives a natural, less static sound.



Hall

Emulates a hall. The parameters are the same as for the Room algorithm above (but the Hall algorithm offers larger Size settings).

Arena

Emulates the ambience in an arena or concert hall, with long pre-delay times (separate for left, right and center):

Parameter	Description
Size	The size of the emulated arena or hall.
Diffusion	At low Diffusion settings, you will hear the individual reverb "bounces" more clearly, while higher settings produce a more "smeared", dense and even reverb.
Left Delay	The predelay time for the left side of the reverb.
Right Delay	The predelay time for the right side of the reverb.
Stereo Level	Adjusts the level of the left and right sides of the reverb. "0" is normal level.
Mono Delay	The predelay time for the mono (center) reverb signal.
Mono Level	Adjusts the level of the mono (center) reverb signal. "0" is normal level.

Plate

A classic plate reverb, excellent for vocals for example. The parameters are:

Parameter	Description
LF Damp	Controls how quickly the low frequencies should decay in the reverb. Raise it to gradually remove low frequencies, making the reverb sound "thinner" and less boomy.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the reverb.

Spring

An emulation of a spring reverb as can be found in guitar amplifiers, organs, etc. The spring reverb has the following parameters:

Parameter	Description
Length	Sets the length of the simulated spring.
Diffusion	At low Diffusion settings, you will hear the individual reverb "bounces" more clearly, while higher settings produce a more "smeared", dense and even reverb.
Disp Freq	When sending a signal to a real-life spring reverb, the initial transient will produce a quick, characteristic sweeping tonal noise. This is because different frequencies in the sound are delayed by different amounts (a phenomenon called dispersion). This parameter controls the frequency of that sound.
LF Damp	Controls how quickly the low frequencies should decay in the reverb. Raise it to gradually remove low frequencies, making the reverb sound "thinner" and less boomy.
Stereo (on/off)	Determines whether the output of the spring reverb should be in mono or stereo.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the early reflections and reverb.
Disp Amount	Sets the amount of dispersion effect (see Disp Freq above).

Echo

This is an advanced echo effect, with diffusion controls and tempo sync. When Echo is selected, the Decay control on the main panel controls the echo feedback (the number of echo repeats). The parameters are:

Parameter	Description
Echo Time	Sets the time between each echo. When Tempo Sync (see below) is off, the echo time is set in milliseconds (10 - 2000 ms); when Tempo Sync is on you set the echo time as a number of 1/16 notes or 1/8 triplet notes, in relation to the current song tempo.
Diffusion	When this is set to 0, the echo will sound as a standard delay with clear, precise repeats. Raising the Diffusion setting will introduce additional echoes very close to the "main" echo repeats, causing a "smeared" echo sound. This will also expand the echo stereo image.
Tempo Sync	Determines whether the echo time should be freely set ("off") or synchronized to Reason's tempo ("on").
LF Damp	Controls how quickly the low frequencies should decay in the echoes. Raise it to gradually remove low frequencies.
Spread	Adjusts the spacing of the additional echoes added by the Diffusion parameter. For a very smeared echo (sound more like a reverb), set both Diffusion and Spread to their maximum values.
Predelay	Sets an additional delay time before the first echo repeat.

Multi Tap

The Multi Tap delay produces up to four different delays with separate delay times, panning and level. The whole set of four delay taps can then be repeated at a given rate. Again, the Decay control on the main panel controls the feedback (the number of repeats for the whole multi tap set). All delay times can be tempo synced.

Note: this algorithm is handled a bit differently since you make separate settings for each delay tap:

- **The parameters to the left of the display are common for all taps.**
- **You use the Edit Select parameter in the top right corner to select which tap to make settings for - the three parameters below affect the currently selected tap.**



Tap 2 selected for editing.

- **You can also set Edit Select to "Repeat Tap" - this is where you specify the repeat time for the whole multi tap "package".**
 With short Repeat times, the first tap may be repeated before the last tap has sounded. This can be used to create very complex multiple delay effects.



The common parameters (to the left) are:

Parameter	Description
Tempo Sync	Determines whether the delay times and repeat times should be freely set ("off") or synchronized to Reason's tempo ("on").

Parameter	Description
Diffusion	Raising the Diffusion setting will introduce additional echoes very close to the “main” repeats, causing a “smeared” delay sound.
LF Damp	Controls how quickly the low frequencies should decay in the echoes. Raise it to gradually remove low frequencies.

When Tap 1 - 4 is selected with the Edit Select parameter, you can make the following settings for the selected delay tap:

Parameter	Description
Tap delay	Sets the delay - the time from the source signal to the tap. When Tempo Sync is off, the delay time is set in milliseconds (10 - 2000 ms); when Tempo Sync is on you set the delay as a number of 1/16 notes or 1/8 triplet notes, in relation to the current song tempo.
Tap level	Adjusts the level of the selected tap.
Tap pan	Adjusts the pan of the selected tap.

When Repeat Tap is selected with the Edit Select parameter, there is only one parameter to the right in the display:

Parameter	Description
Repeat Time	Sets the time between each repeat of the whole multi tap set. The number of repeats is set with the Decay control on the main panel. When Tempo Sync is off, the repeat time is set in milliseconds (10 - 2000 ms); when Tempo Sync is on you set the repeat time as a number of 1/16 notes or 1/8 triplet notes, in relation to the current song tempo.

Reverse

The Reverse reverb algorithm in RV7000 Mk II is special in that it actually “moves” the source audio as well. Sounds fed into the Reverse reverb are “sampled”, a reverse reverb is created and played back and finally the “sampled” original sound is played back. For example, if you feed a snare drum hit into the Reverse reverb, you will hear a rising “backwards” reverb, followed by the snare drum hit.

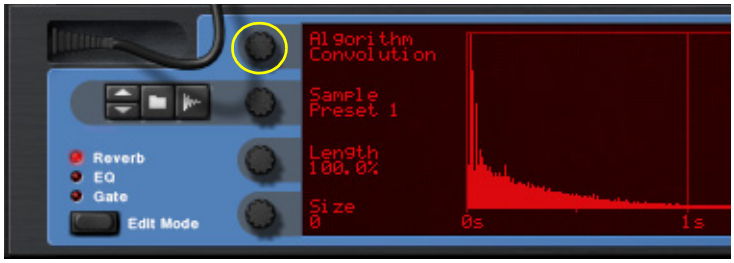
Therefore, you probably don’t want to hear the first, original (dry) sound. Here is how to set this up:

→ **Connect the RV7000 Mk II as an insert effect and make sure the Dry/Wet control on the main panel is set fully to “Wet”.**

Note that with this algorithm, raising the Decay setting on the main panel will make the reverse reverb start earlier and build up under a longer time. Similarly, the HF Damp parameter affects how fast the high frequencies are built up in the reverse reverb. In the remote panel, the Reverse algorithm has the following parameters:

Parameter	Description
Length	This sets the time from when the source signal is fed into the reverb until it is played back again. It is during this time you will hear the reverse reverb, as shown in the display. The time can be set in milliseconds or as note values, depending on whether Tempo Sync is off or on. Note: As stated above, the Decay setting determines the length of the actual reverse reverb - in essence how soon it starts after the source signal. But of course, the reverse reverb cannot start *before* the original source signal! If you set Decay to a longer time than the Length setting, the reverse reverb will start abruptly, immediately when the source signal is fed into the reverb. If this sounds complicated, just take a look at the RV7000 Mk II display and try the settings - you will soon see how it works. Note also that very high Length settings demand a lot of processor power. This can be reduced by adjusting the Density parameter, see below.
Density	Density governs the “thickness” of the Reverse effect. If this parameter is turned down to zero, the effect produces individual delays rather than a dense “wash”, which can be used as a special effect. Worth noting is that if Density is set to around 50%, this can considerably reduce the CPU load without altering the sound of the effect too much. Exactly how much the Density parameter can be reduced without altering the sound depends on the source material.
Rev Dry/Wet	Sets the balance between the “moved” source signal (“dry”, low values) and the reverse reverb (“wet”, high values).
Tempo Sync	Determines whether the Length setting should be freely set (“off”) or synchronized to Reason’s tempo (“on”).

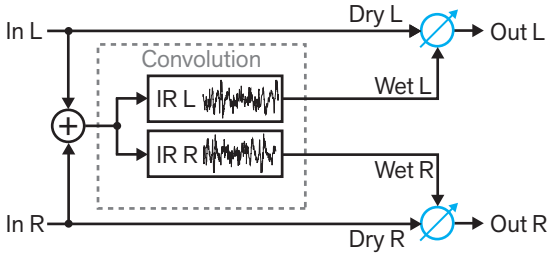
Convolution



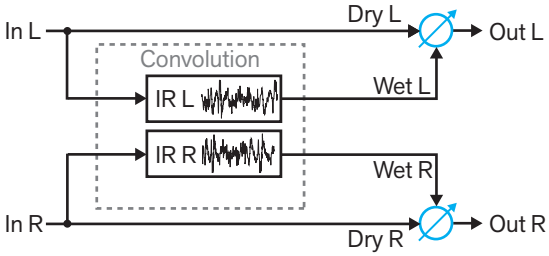
The zero-latency Convolution algorithm uses impulse response samples to generate effects. Basically, "convolution effects" are the results of multiplying the frequency spectra of the input signals with the frequency spectra of impulse response samples, and thus generating a signal with the "character" of the impulse response sample. If the impulse response sample is a recording of the reflections of a large room, for example, the resulting effect will be "the input audio signal played back in a large room".

RV7000 Mk II comes with three built-in preset impulse response samples. You can also use any other samples for the convolution algorithm, to generate all kinds of reverbs and special effects. You can even sample your own impulse responses and use in the convolution algorithm in RV7000 Mk II.

Note that in the Convolution algorithm the input signals for the effect are first summed to mono (except in Parallel Stereo Mode (see "Stereo Mode")) and then processed with the impulse response sample. The figures below shows the signal routings in the Convolution algorithm:



(All Stereo and Mono modes)



(Parallel mode)

In the remote panel, the Convolution algorithm has the following parameters:

Parameter	Description
Sample Preset	Here you select one of the preset impulse response samples - plus your own sample (if you use that, see "Loading impulse response samples").
Length	Sets the length (end point) of the currently used impulse response sample.
Size	Simulates the "size" of the impulse response sample, in practice its pitch, in semitone steps. -12 means pitching up the impulse response sample 1 octave and 12 means pitching it down 1 octave.
LF Damp	Controls how quickly the low frequencies should decay. Raise it to gradually remove low frequencies, making the effect sound "thinner" and less boomy.

Parameter	Description
Stereo Mode	Determines how the output signal should be presented. Note that the stereo effect is derived from the impulse response sample itself. So, the wider the stereo effect in the impulse response sample, the wider the stereo effect in the output. Note that if you use a mono impulse response sample, all effects below will be in mono! Stereo: The input signals are summed and then a stereo effect with a spread of 100% is applied. Stereo75%: The input signals are summed and then a stereo effect with a spread of 75% is applied. Stereo50%: The input signals are summed and then a stereo effect with a spread of 50% is applied. Stereo25%: The input signals are summed and then a stereo effect with a spread of 25% is applied. Mono: The input signals are summed and then the effect is panned to a centered mono signal (0% spread). M->S Slow: From Mono slowly panned out to full Stereo. M->S Fast: From Mono quickly panned out to full Stereo. S->M Slow: From full Stereo slowly narrowed down to Mono. S->M Fast: From full Stereo quickly narrowed down to Mono. Parallel: Individual processing of the two L and R stereo input signals (dual mono).
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the convolution effect. A negative predelay masks the start of the sample, so if the sample has unwanted initial transients or silence, these can be "removed".
Gain	Sets the amplification or attenuation of the effect signal.

Below are some things to keep in mind when you are working with the Convolution algorithm:

- **If you want the impulse response sample to play back exactly like the original, make sure the Decay knob is at max, the LF Damp knob at zero and the Hi EQ knob at its 12 o'clock position. Also, make sure the Length parameter is at 100% and the Size parameter at 0.**
- **The Length value of the impulse response sample is also affected by the Decay knob setting.**
 If the impulse response sample is quiet at the end, reduce the Length value to cut it off a little earlier. The Decay parameter introduces a smoother "cutoff" at the end, which might be desirable in many situations.
- **Changing some convolution parameters re-calculates the impulse response in real time. Therefore, modulating these parameters might give unexpected results. Specifically:**

 Using CV to modulate the RV7000 Mk II in Convolution mode is not recommended.

 If you're using the Gate function (see "The Gate section") in combination with Convolution, we recommend setting the Decay Mod parameter to 0 (see "Decay Mod").

Loading impulse response samples

Besides the preset impulse response samples, you can load any other samples into the RV7000 Mk II and use as impulse responses. Loading a sample automatically switches the Algorithm to Convolution.

→ **Drag and drop a sample onto the RV7000 Mk II panel to automatically load it in the Convolution Algorithm.**

Or, do as follows:

1. **Click the Browse Samples button on the Programmer panel to set browse focus to the RV7000 Mk II.**



2. **Select the desired sample in the Browser and load it.**



Here we have selected and loaded the "Fx_DubHead.WAV" sample to use as impulse response. You can see that the sample name is now displayed to the right of the Preset knob.

! **Note that the maximum length of a sample used as an impulse response is approximately 12 seconds. Longer samples are automatically truncated to 12 seconds (non-destructive).**

! **Note that any embedded loop data in the samples are disregarded when loaded into the RV7000 Mk II!**

! **Note that the Programmer display only shows the first 4 seconds of longer samples.**

► **To achieve stereo effects in the Stereo Mode alternatives (see "Stereo Mode") you have to use stereo samples.**

3. **Edit the Convolution parameters on the RV7000 Mk II panel until you are satisfied.**

→ **If you want to save your RV7000 Mk II patch with your impulse response sample, click the Save Patch button on the upper panel:**



! **Note that the impulse response sample is NOT saved in the patch itself - only a reference to the sample! If you have loaded an external sample (that is not in the Factory Soundbank), be sure NOT to move/remove the sample from its original location on your computer.**

The EQ section



The equalizer in RV7000 Mk II affects the wet reverb sound only and is used for shaping the character of the reverb. There are two EQ bands, one for low frequencies (shelving) and one full-range parametric EQ.

- **To activate the EQ, click the EQ Enable button on the main panel so that the indicator lights up.**
- **To make EQ settings, select "EQ" with the Edit Mode button to the left in the remote programmer panel.**
- **In this mode, the remote programmer display shows a frequency curve, indicating the settings you make with the EQ parameters.**

The parameters are:

Parameter	Description
Low Gain	The amount of cut or boost of the low-shelving filter.
Low Freq	The frequency below which the Low Gain cut or boost is applied.
Param Gain	The amount of cut or boost for the parametric EQ.
Param Freq	The center frequency of the parametric EQ, e.g. at which frequency the level should be decreased or increased.
Param Q	This governs the width of the affected area around the set center frequency. The higher the value, the narrower the affected frequency range.

- **Remember that you have a third EQ band at your disposal - the HI EQ parameter on the main panel.**
 The reason why this is on the main panel and not in the EQ section is simply that it's a setting you may want to adjust often, without having to open the remote programmer panel.

The Gate section



The Gate section allows you to create gated reverb effects with a lot of options and possibilities. You can either trigger the gate from the source audio signal or via MIDI or CV.

When triggering the gate from the source audio signal, it works like this:

- **The gate "listens" to the source (dry) signal and opens whenever the signal reaches a certain threshold level.**
- **The reverb sound is sent through the gate - when the gate is closed you won't hear the reverb.**
- **When the source signal level drops below the threshold level, the gate closes after a time that depends on the Hold parameter and the level of the source signal (see the parameter table).**

→ **If you need the gate to be open for an exact duration (time), you should trigger it via MIDI or CV.**
In audio trigger mode, the actual gate time will vary depending on the source signal.

When triggering the gate via MIDI or CV, it works like this:

- **The reverb sound is sent through the gate - when the gate is closed you won't hear the reverb.**
- **Whenever the gate receives any MIDI note (sent to the RV7000 Mk II) or a gate signal (connected to the Gate Trig CV input on the back of the RV7000 Mk II), the gate opens for the duration of the note or gate signal.**

Note:

→ **To activate the Gate, click the Gate Enable button on the main panel so that the indicator lights up.**

→ **To make Gate settings, select "Gate" with the Edit Mode button to the left in the remote programmer panel.**

→ **In this mode, the remote programmer display shows two meters - one showing the signal level (with an indication of the threshold level) and one showing the status of the gate.**
This is useful for checking what happens, how the gate triggers, etc.

The parameters for the Gate section are:

Parameter	Description
Threshold	When Trig Source is set to “Audio”, this determines the audio signal level at which the gate opens. If you raise this setting, only very loud sounds will open the gate.
Decay Mod	This modulates the reverb Decay parameter so that the decay time is lowered when the gate closes. When this is set to zero, no decay modulation happens - this means that if the gate is closed and then opened again, you may hear “previous” reverb tails that are still ringing. If you raise the Decay Mod setting, the decay will automatically be lowered when the gate is closed, eliminating this effect.
Trig Source	Determines whether the gate should be triggered by audio or MIDI/CV, as described above.
High Pass	A high-pass filter that affects the audio that triggers the gate (only active when Trig Source is set to “Audio”). If you raise this setting, sounds with low frequencies only will not open the gate. Note that this setting doesn’t affect the sound of the reverb, only the triggering mechanism.
Attack	Determines how long it takes for the gate to open after a triggering signal has been received.
Hold	This parameter is only active when Trig Source is set to “Audio”. Hold affects how quickly the gate closes, in the following way: Internally, the gate is controlled by an envelope follower that analyzes the source signal level and generates a “level CV signal” accordingly. This signal is compared to the Threshold level to determine whether the gate should be opened or closed. The Hold parameter affects how quickly the envelope follower responds when the source signal level drops - you could say that this is the decay control for the envelope follower. The higher the Hold setting, the longer it will take for the envelope follower signal to drop below the threshold level and close the gate. But the resulting time also depends on the source signal level - with a loud signal, it will take longer time for the envelope follower to drop to the threshold level. Therefore, the actual gate time depends both on the Hold setting and on the character of the source audio.
Release	Determines how long it takes for the gate to close after the Hold time.

CV Inputs



On the back of the RV7000 Mk II you find three CV inputs. These are:

Parameter	Description
Decay	Controls the reverb decay or echo/delay feedback via CV.
HF Damp	Controls the HF Damp parameter on the main panel.
Gate Trig	Used for triggering the Gate section with a gate signal. The length of the gate signal determines the length of the gated reverb.

! Using the CV inputs to modulate the RV7000 Mk II in Convolution mode is not recommended.

Chapter 37
Neptune Pitch Adjuster and Voice Synth

Introduction



The Neptune Pitch Adjuster and Voice Synth device is a combined monophonic vocal pitch corrector, pitch shifter and polyphonic voice synth. Neptune was designed with focus on high-quality vocal processing but can also be used on other material. However, due to the signal characteristics of other types of audio (complex inharmonic instrument sounds, polyphonic material etc.) the result of the pitch adjustments might not be what you would expect. Don't hesitate to experiment, though!

Used as a pitch corrector, Neptune can automatically correct flat monophonic input signals and output corrected notes in real-time. The pitch correction can be adjusted from totally transparent to hard, robot-like. The correction can be controlled from predefined scales, from input MIDI notes or from a combination of both!

When used as a pitch shifter, Neptune performs overall pitch-shifting of incoming monophonic audio in real-time and transposes the output to a defined value within a ±1 octave range.

The Voice Synth section in Neptune allows you to add additional voices - harmonies - to a monophonic vocal input signal and to control the voices via MIDI - from sequencer notes or by playing on your MIDI master keyboard.

Neptune also features controls for adjusting formants and for adding pitch bend and vibrato to the processed signal.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

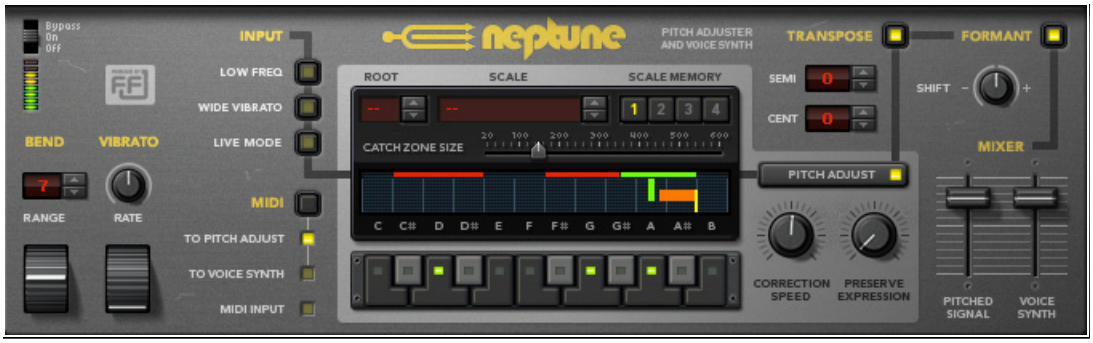
Typical use cases

Neptune can be used in a variety of ways and situations. Here are some examples:

- **Pitch correction of a vocal track**
 Neptune is used as an Insert Effect on the vocal audio track, with the Pitch Adjust's Scale set to desired scale. Set the Correction Speed parameter to a moderate value for transparent pitch correction - or set the Correction Speed parameter to max value and the Preserve Expression parameter to minimum for a hard "robotic" character.
- **Octave dub**
 Neptune is used as a Send Effect. The Pitch Adjust section is turned off and the Transpose section is activated. The Semi parameter is set to +12 to transpose up 1 octave. The Send FX control on the mixer is used for balancing the level of the octave-shifted voice. The Formant Shift control is used for making the transposed voice sound more natural (or unnatural, if that's desired).
- **Create backing harmonies on a vocal track**
 Neptune is used as an Insert Effect on the vocal audio track and MIDI Input is routed to the Voice Synth via the Neptune sequencer track. Play some notes on your MIDI master keyboard and use the Voice Synth breakout jacks connected to a separate Mix Channel for further processing.
- **Pitch shift drums (non-pitched input)**
 The Pitch Adjust section is turned off and the Transpose section is activated. The Transpose parameters are set to desired values.

Overview and basic concepts

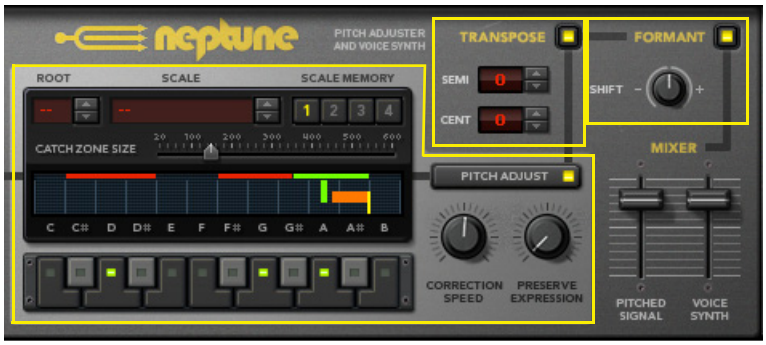
Sections overview



The Neptune panel is divided into three main sections that are connected in series:

- **The Pitch Adjust section.**
 Here is where you control the pitch correction settings, including scale, correction speed etc. Here is also where you find the big display where you can view Input Pitch, Target Note and Catch Zone etc. See "Using pitch correction". for more details.
- **The Transpose section.**
 Here is where you control the pitch shifting parameters. See "Using pitch shifting (Transpose)" for details.
- **The Formant section.**
 Here is where you can control and adjust formants. See "Using Formant control" for details.

These three main sections can be used separately, or in combinations, by clicking the ON/Off button of the respective section:



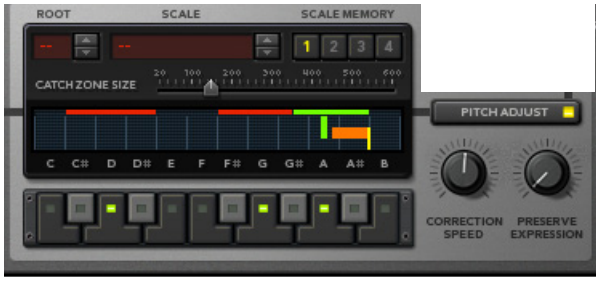
The first section in the signal chain is the Pitch Adjust section. To get a realistic result out of the pitch correction, you will probably want to use it in combination with the Formant section - especially if you control the target pitch via MIDI and the target pitch differs a lot from the input pitch.

The Transpose section can be used stand-alone when you only want to perform stiff pitch shifting of the input audio. If you want to pitch-shift atonal audio, you should use the Transpose section stand-alone, or in combination with the Formant section, for best result. You can also use the Transpose section together with the Pitch Adjust section for transposing pitch corrected audio by a defined amount.

The Formant section can be used stand-alone if you only want to change the character of the input audio - without affecting the pitch. Used on vocals or speech, the Shift knob lets you control the "gender" of the voice.

Besides these three main sections, there are also additional parameters for defining audio input characteristics (see "Input signal type"), MIDI control (see "MIDI Input") and output level control (see "The Output Mixer section").

The display



The big display in the center of the Neptune panel shows the following information:

- **The Input Pitch of the incoming audio signal.**
 The detected Input Pitch is displayed as a yellow vertical line above the keyboard.
- **The Target Note to which Neptune corrects the output signal.**
 The Target Note is displayed as a green vertical line above selected notes on the keyboard.
- **An orange horizontal line between the Input Pitch and the Target Note.**
 The orange line shows the distance and direction from Input Pitch to Target Note.
- **The Catch Zones, i.e. the pitch "window" which determines to what Target Note to correct the Input Pitch.**
 The Catch Zones are displayed as red horizontal boxes above each selected key on the virtual keyboard. As soon as a detected Input Pitch lies within a Catch Zone, the Catch Zone box above the corresponding Target Note switches to green. See "Setting Catch Zone Size".
- **The virtual keyboard, where you can select your own notes or custom scale to correct the audio to.**
 See "Setting Root Key and Scale".

Setting up for pitch processing

The most flexible method of performing pitch adjustments is to apply it to already recorded audio tracks in the sequencer. Doing so will give you total freedom to edit and change the pitch adjustment settings afterwards without needing to re-record any audio.

To set up Neptune for pitch adjustments of recorded audio tracks, proceed as follows:

1. **Create a Neptune device and use the Reason Rack Plugin instance as an insert effect on the audio track.**
2. **If you want to control the pitch adjustment via your MIDI keyboard, route MIDI to the Neptune device from your DAW.**

Now, you are all set for pitch adjustments of the audio track. For information about pitch correction, see "Using pitch correction", for information about pitch-shifting, see "Using pitch shifting (Transpose)", and for info about creating additional vocal harmonies, see "Using the Voice Synth".

Using pitch correction

Pitch correction is great if you, for example, have a vocal track in your song that contains flat notes. Instead of having to overdub the flat parts on the audio track, you can automatically adjust the pitches of the flat notes using Neptune. You can also sing through the Neptune in real-time and get a beautifully pitched voice!

There are two basic ways of using pitch correction in Neptune:

- **Automatic pitch correction using Scales, see "Using automatic pitch correction".**
- **Manual pitch correction using MIDI note input, see "Using manual pitch correction".**

You can also combine the two methods if you like. First of all, though, you will need to make some basic settings:

Basic settings for pitch correction

Proceed as follows to set up the panel parameters for pitch correction:

1. **Set up the Neptune device as desired according to the descriptions in "Setting up for pitch processing".**
2. **Make sure the "Pitch Adjust" button is activated and that the "Transpose" button is off.**



- **Whether the Formant button should be on or off depends on the situation.**
 For very small pitch corrections, you might want to leave this off; for more substantial pitch changes you will get the most natural sound when this is activated.

3. **Define the Input signal type by using the "Low Freq", "Wide Vibrato" and "Live Mode" buttons:**



- **Activate "Low Freq" for low-frequency material such as a deep voice etc.**
 The "Low Freq" mode will make Neptune detect low frequency notes in a more precise way. Note that the latency will become longer due to the fact that low frequencies have longer cycle times.
- **If the input voice contains a lot of vibrato, it might be a good suggestion to activate the "Wide Vibrato" button.**
 If your input audio has a heavy vibrato, this can cause Neptune to detect the wrong pitches. The result can be wobbling notes, unwanted swoops and glides etc. Activating the Wide Vibrato button will make the pitch detection ignore any vibrato in the input audio, eliminating the problems. Note however that the vibrato can still be retained in the processed sound, by raising the Preserve Expression parameter (see "About the Preserve Expression parameter").
- **If you are going to use pitch correction by singing through Neptune in real-time, try activating the "Live Mode" button.**
 This will reduce the latency of the pitch correction to a minimum, which might be preferable if you want to monitor the corrected signal as you sing. Note, however, that the audio quality may be lower in Live Mode.

4. **Make sure the "MIDI" button is set to "To Pitch Adjust".**



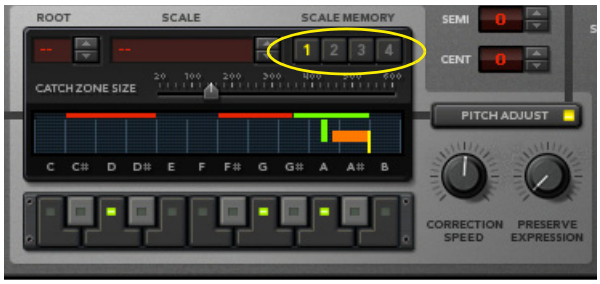
This will enable the pitch correction to be controlled also via MIDI (see "Using manual pitch correction").

Using automatic pitch correction

With the Root and Scale functions you can define what notes you want the incoming audio to be corrected to. The input audio will then be automatically corrected to the defined scale, without further user interaction. With the Catch Zone, Correction Speed and Preserve Expression parameters you can set the character of the adjusted signal.

- **It's also possible to manually override the automatic settings via MIDI at any time, see "Using manual pitch correction".**

About the Scale Memory



There are four Scale Memory slots in the Pitch Adjust section. The purpose of Scale Memories is to allow for automation of different Root Key, Scale and Catch Zone settings. A selected Scale Memory slot automatically stores which Root Key and Scale notes are active (see "Setting Root Key and Scale"), just like a Redrum stores what 16th note buttons are selected in a pattern. The Scale Memory slots also store Catch Zone settings (see "Setting Catch Zone Size"). By using automation of the Scale Memory slots from the Neptune sequencer track, you can easily switch characteristics throughout the song. The Scale Memory settings are automatically saved with the document when you save your Song.

! **Before you perform any edits of the Root Key, Scale and/or Catch Zone parameters, make sure you have selected a Scale Memory slot that you want to overwrite.**

Setting Root Key and Scale

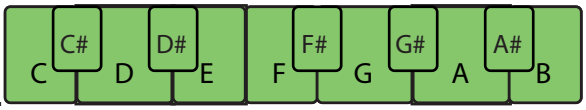
! **Any Root Key and/or Scale settings changes you make are automatically stored in the currently selected Scale Memory slot in real-time, see "About the Scale Memory".**

Define the Root Key and Scale parameters as follows:

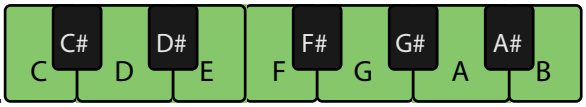
1. **Set the Root Key with the Root spin controls.**
 The key can be chosen between C and B in a one octave span, covering each key in the western 12-tone scale.



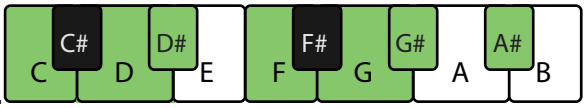
2. **Set desired scale with the Scale spin controls.**
 The Scale parameter can be set to any of the following preset scales, as indicated in the Scale display:



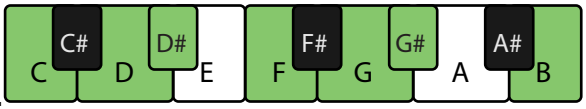
Chromatic



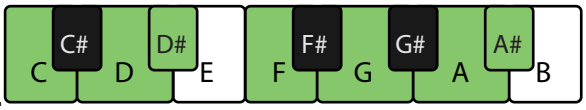
Major with C as Root Key



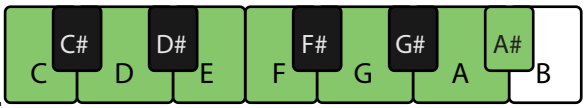
Natural Minor with C as Root Key



Harmonic Minor with C as Root Key



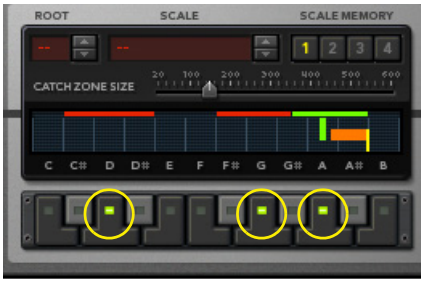
Dorian with C as Root Key



Mixolydian with C as Root Key

! *** When Chromatic is selected, the Root display will switch to show "--" since all notes in the 12-tone scale are included and the root key is of no importance here.**

→ **You can also define your own custom scale by clicking on the keys in the display.**
Selected notes in your custom scale will be indicated with green LEDs:



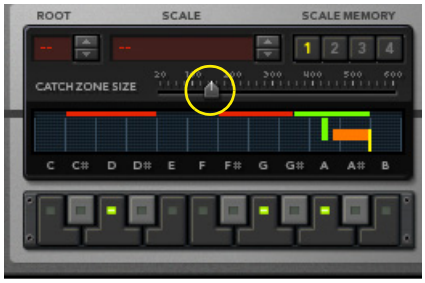
A custom scale containing the notes D, G and A.

When you set your own scale both the Root Key and Scale displays will switch to show "--".

- **The set Key and Scale is automatically repeated for every octave throughout the entire note range.**

! **Note that this doesn't necessarily need to be an actual scale - sometimes you may only need to activate one or two notes, passing the other notes through, uncorrected.**

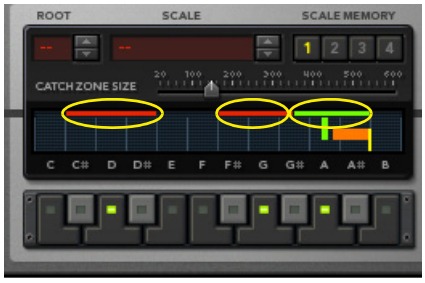
Setting Catch Zone Size



The Catch Zone Size parameter defines what pitches in each octave should be "caught" and adjusted towards the closest Target Notes.

! **Any Catch Zone Size settings changes you make are automatically stored in the currently selected Scale Memory slot in real-time, see "About the Scale Memory".**

The Catch Zones are shown as boxes above each selected note in the keyboard display:

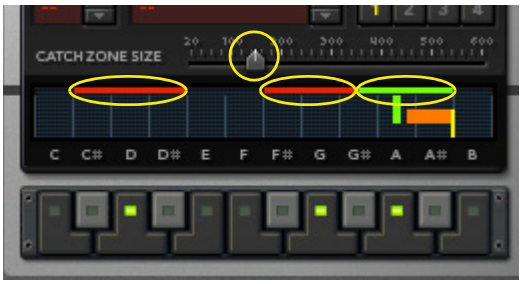


The Catch Zones for the notes in a custom scale with notes D, G and A selected.

- **The Catch Zone size is set with the Catch Zone Size knob.**
The range is ±20 to ±600 cent, set in 20 cent steps. The range is always centered around the selected notes in the scale and the default value is ±100 cents.

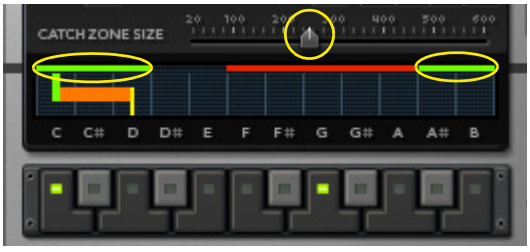
! **Pitches outside or in-between Catch Zones are not caught and adjusted, but let through unprocessed.**

- **Catch Zones cannot overlap each other. If the Catch Zone Size is increased so that adjacent zones touch each other, they will stop expanding in that direction and meet "half ways".**
 The Catch Zone size in the picture below is set to ±150 cents. As you can see, the Catch Zones for notes G and A have "collided" and therefore met half ways in the center of the G# note. If the Catch Zone size should be increased further, the G Catch Zone would expand only downwards and the A Catch Zone only upwards.



The Catch Zones for the selected notes D, G and A in a custom scale.

- **If the Catch Zones should extend on either side of the 12-note keyboard range, they will "wrap around".**
 In the picture below, notes C and G are selected in a custom scale. The Catch Zone Size is set to ±250 cents. Since the Catch Zone for the C note extends also to the left outside the display, the Catch Zone "wraps around" visually and continues from the B note down to the A# note. Since the Scale is repeated downwards and upwards for every octave, the result of this setting is that the Catch Zone for the C note will cover the B and A# notes in every octave.



The Catch Zones for the selected notes C and G in a custom scale, with the C note Catch Zone "wrapped around" to cover also notes B and A#.

Setting Correction Speed



The Correction Speed parameter controls how fast the Input Pitch should be adjusted to the Target Note. The range is from Slow (knob turned fully counter-clockwise) to Fast (knob turned fully clock-wise).

- **For a natural, transparent correction, a setting around the 12 o'clock position is ideal in most situations.**
- **A very fast correction speed will create almost a "stepped" correction.**
 This is the setting of choice for creating the infamous "robot voice" effect known from numerous radio hits.
- **A slow correction speed will make the pitch correction almost unnoticeable during fast passages in the music.**
 This is because the correction won't have time to set in before new incoming pitches are detected.

About the Preserve Expression parameter



The Preserve Expression parameter controls how much vibrato in the input audio should be let through when you use a fast Correction Speed setting (see "Setting Correction Speed").

- **With a minimum Preserve Expression value and a fast Correction Speed there will be almost no natural vibrato left in the pitch corrected voice.**
- **With max Preserve Expression value and a fast Correction Speed the original vibrato is still preserved.**
- **With max Preserve Expression value and a slow Correction Speed the original audio is let through almost unaffected.**

About adding Pitch Bend and Vibrato

It's also possible to add Pitch Bend and/or Vibrato via MIDI while using the automatic pitch correction:

→ **Select "To Pitch Adjust" in the MIDI Input section.**



This allows for incoming MIDI Pitch Bend and Vibrato (Mod Wheel) data to control the output pitch and to add vibrato to the output signal.

- **Clicking and moving the wheels on the panel will also generate pitch bend and vibrato.**

Using manual pitch correction

In Neptune it's possible to override the automatic (scale) pitch correction settings with monophonic MIDI note data at any time. This makes it possible to control the desired output pitches either by playing live on a MIDI master keyboard - or by playing back recorded MIDI notes from the Neptune track in the sequencer!

1. **Add a Reason Rack Plugin instance and create a Neptune device.**
2. **To allow for MIDI note control of pitch correction, select "To Pitch Adjust" in the MIDI Input section:**



- **When you sing through Neptune and hold down a key on your MIDI master keyboard, the output pitch will correspond to the held MIDI note.**
 As soon as any MIDI note is received by Neptune, the Root Key, Scale and Catch Zone settings will be temporarily ignored. However, the Correction Speed (see "Setting Correction Speed") and Preserve Expression parameters (see "About the Preserve Expression parameter") will still be active.

! **Note that MIDI control of the pitch is monophonic.**

- **When pitch correction is controlled via MIDI, the corrected audio signal will also respond to Pitch Bend and Vibrato modulation.**

! **If you transpose pitches several semitones compared to the input pitch, there could be strange formant effects that may or may not be desirable. To make heavily pitched signals sound more natural, you can use the formant correction function described in "Using Formant control".**

! **Note that MIDI control of the pitch correction will only momentarily override the Root Key, Scale and Catch Zone settings. As soon as there is no MIDI Note present, the automatic (scale) pitch correction settings will take over again.**

► **It's also possible to generate polyphonic voices (harmonies) via MIDI using the Voice Synth function, see "Using the Voice Synth".**

Using pitch shifting (Transpose)

Pitch shifting basically means altering the pitch of the input audio by a set factor - the transpose value - and output the result. As opposed to pitch correction, pitch shifting can be made on any type of monophonic audio input signal; it doesn't even have to have a regular pitch (tone). Pitch shifting can be used to generate a number of cool effects, for example:

- **Voice doubling effects by using slight transposition of only 5-10 cents.**
- **Adding an additional voice one octave up or down from the original lead vocal.**
- **Creating dark spooky monster voices by transposing speech down heavily.**
- **Creating chipmunk effects by transposing speech up heavily.**

The list goes on...

Use the Transpose function as follows:

1. **Click the Transpose button to activate the Transpose section.**



2. **Adjust the transposition using the Semi and Cent spin controls in the Transpose section.**
 The Transpose range is ±12 semitones with a fine tuning range of ±50 cents.

→ **If you want to transpose pitch corrected signals, make sure the Pitch Adjust section is on and has the desired parameter settings.**

→ **If you want to transpose non-pitched signals, such as speech, switch off the Pitch Adjust section.**

Using Formant control

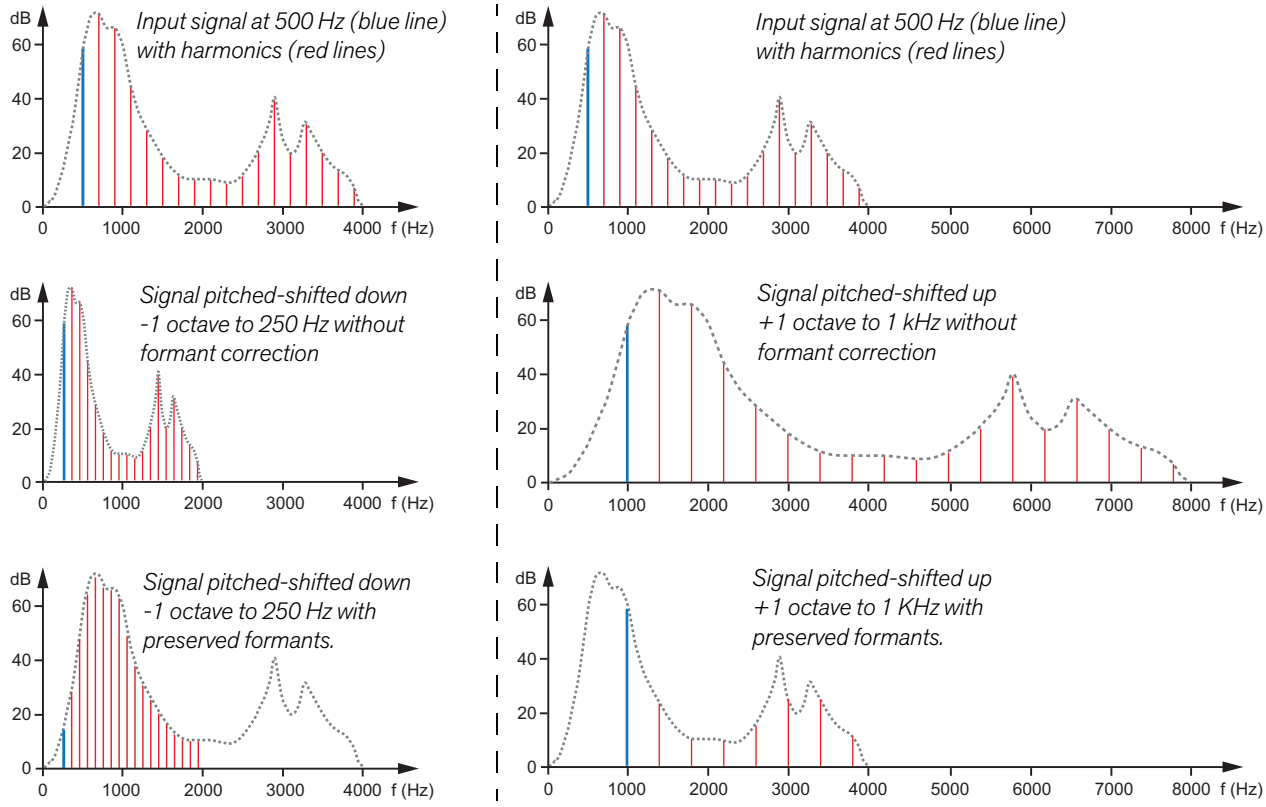
What are formants?

Formants can be described as a sonic "footprint" of an acoustic space. The practical effect of formants could be compared to a multi-peak filter acting on the frequencies in a sound. An acoustic guitar, for example, has a body shape which makes is sound the way it does. The same goes for a human vocal tract (throat and mouth cavity); every human vocal tract has a unique "shape" which gives the voice its character. It is these shapes that produce formants. A big difference between a vocal tract and a guitar body, though, is that the vocal tract changes shape as you sing different vowels. This also means that the formants will change.

When you pitch shift a signal up or down, the formant "multi-peak filter" will move up or down with the signal frequency (just like a traditional synthesizer filter would with full keyboard tracking activated). The result will be a signal that not only is pitched but also changes character. In some situations this might be what you want, but when it comes to pitch shifting vocals you will probably often want the pitch-shifted signal to sound like it's sung by the same person. Therefore, Neptune features a formant control function.

The Formant section in Neptune lets you control the formants so they don't move along with the pitched signals. Neptune continuously samples and analyses the input audio and determines both the pitch and the current formants of the signal in real-time. The formants are then automatically applied to the pitch-adjusted output signal in real-time.

The picture below shows schematic examples of a 500 Hz signal, pitch-shifted down and up one octave respectively, without and with formant correction. The formants are the peaks of the gray dotted lines in the graphs:



Pitch-shifting without and with formant correction applied. The left column shows -1 octave pitch-shifting and the right column +1 octave.

Using the Formant function

When the Formant section is not active, the formants will move along with the pitched signal. On vocals this could generate unnatural "gender change" effects, especially on heavily pitched voices. If this is not the desired effect, you should activate the Formant section:

1. **Activate the Formant function by clicking the Formant button.**



2. **Adjust the Formant shift with the Shift knob.**
 The Shift range is ±1 octave.

- **The default 12 o'clock position means that the formants will be "locked" to the input signal and won't move with the shifted pitches.**
 Used on vocals this will give the impression of the same person singing at the adjusted pitches. (Refer to the picture in the "What are formants?" section for graphical examples of this particular Shift setting.)
- **Turned further counter-clockwise, the formants will be "locked" to the input signal and displaced downwards relative to the input signal.**
 Used on vocals this will give the impression of a more deep male character voice.
- **Turned further clockwise, the formants will be "locked" to the input signal and displaced upwards relative to the input signal.**
 Used on vocals this will give the impression of a more soprano/female/child character voice.

! **Note that Voice Synth has its own separate "built-in" formant correction and is not affected by the Formant section settings.**

Using the Voice Synth

The Voice Synth function lets you use Neptune as a harmony processor to generate polyphonic voices out of a monophonic input signal. The application is similar to how you would use a vocoder, such as the BV512 Vocoder, for example. However, using the Voice Synth in Neptune will produce completely transparent and natural sounding harmonies with pitches controlled from MIDI Note data.

1. **Add a Reason Rack Plugin instance and create a Neptune device.**
2. **Click the MIDI radio button to select "To Voice Synth" in the "MIDI" section. Also make sure the Voice Synth volume slider is raised:**



- **When you sing through Neptune and hold down a note or a chord on your MIDI master keyboard, the output pitch(es) will correspond to the held MIDI note(s).**
- **The Voice Synth harmonies will also respond to any Pitch Bend and Vibrato modulation.**

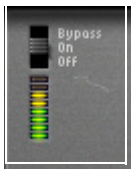
! **The parameters in the Pitch Adjust, Transpose and Formant sections are ignored by the Voice Synth.**

► **You can mix Pitched Signal with the Voice Synth signal in the Mixer section. If you only want the Voice Synth sound, lower the Pitched Signal fader.**

Panel parameters

On the front panel of Neptune, you will find the following parameters and displays:

Level Meter and Bypass/On/Off switch



Parameter	Description
Bypass/On/Off switch	In Bypass mode, the input signal passes through unaffected to the main outputs of the device. The separate Voice Synth outputs are automatically muted. In On mode, the device outputs the processed signal. If the Voice Synth is used, its output signal are routed to the separate Voice Synth outputs. Off mode mutes the inputs, silencing the device.
Level meter	Shows the input signal level.

Bend and Vibrato wheels

! **Choose what the Bend and Vibrato wheels should control by selecting desired destination in the MIDI Input section - see "MIDI Input".**



- **The Pitch Bend wheel is used for bending the pitch of notes, much like bending the strings on a guitar or other string instrument.**
- **The Vibrato wheel can be used for applying vibrato to pitch corrected signals or to the Voice Synth harmonies.**

Most MIDI keyboards have Pitch Bend and Modulation controls. Use these to control pitch bend and vibrato, or use the wheel controls on the panel by clicking and moving the mouse.

Bend Range

The Range parameter sets the maximum amount of pitch bend, i.e. how much it is possible to change the pitch by turning the wheel fully up or down. The maximum range is ±12 semitones (±1 octave). You change the value by clicking the spin controls above of the display.

Vibrato Rate

Use the Rate knob to set the rate of the vibrato controlled by the Vibrato wheel.

Input signal type



Here, you define the characteristics of the input signals you use.

Low Freq

Activating Low Freq mode will ensure more accurate tracking of low-frequency audio signals, such as bass voices. Low Freq mode is suitable for input frequencies below the note F1, which is approximately 44 Hz. The detection in Low Freq mode will work down to approximately 22 Hz which corresponds to note F0.

! **Note that the pitch detection latency will be somewhat longer in Low Freq mode due to the longer cycles times of low frequency signals.**

Wide Vibrato

If your input audio has a heavy vibrato, this can cause Neptune to detect the wrong pitches. The result can be wobbling notes, unwanted swoops and glides etc. Activating the Wide Vibrato button will make the pitch detection ignore any vibrato in the input audio, eliminating the problems.

Note however that the vibrato can still be retained in the processed sound, by raising the Preserve Expression parameter (see "About the Preserve Expression parameter").

Live Mode

This is an ultra-fast tracking mode, perfect for when you want to monitor your pitch adjusted signal in real-time. However, we recommend that you turn this off when it's time to play back and mix your recordings, for the highest audio quality.

MIDI Input



The MIDI Input section features a radio button which allows you to route incoming MIDI to either of the following destinations:

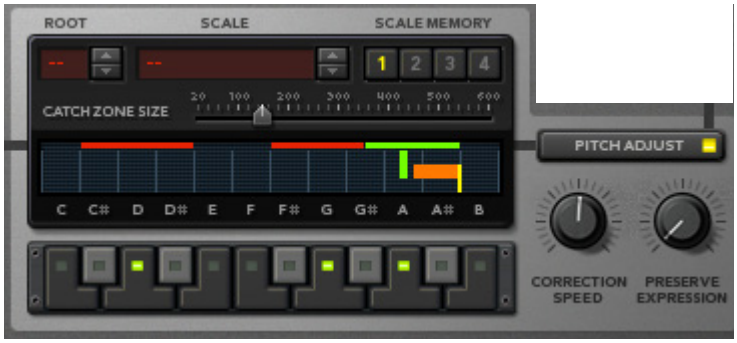
Pitch Adjust

Routes incoming MIDI Note data to the Pitch Adjust section for manual control of pitch correction, see "Using manual pitch correction". Pitch Bend and Vibrato (Mod Wheel) MIDI data will also be routed to the Pitch Adjust section.

Voice Synth

Routes incoming MIDI Note data to the Voice Synth for generating additional harmony voices, see "Using the Voice Synth". Pitch Bend and Vibrato (Mod Wheel) MIDI data will also be routed to the Voice Synth.

Pitch Adjust section



Pitch Adjust section On/Off

Click the Pitch Adjust button to activate/deactivate the Pitch Adjust section.

Root, Scale and Keyboard

Set the Root key and Scale with the spin controls. You can also create your own custom scales by selecting individual notes in the Keyboard display. See "Setting Root Key and Scale" for more details on how to use these functions.

Scale Memory

The purpose of Scale Memories is to allow for automation of different Root Key, Scale and Catch Zone settings. A selected Scale Memory slot automatically stores which Root Key and Scale notes are active. The Scale Memory slots also store Catch Zone settings.

The edits you make of the Root, Scale and Catch Zone parameters are automatically stored in the selected Scale Memory slot. The settings of the Scale Memory are saved together with the rest of the Song data. Refer to "About the Scale Memory" for details on how to use the Scale Memory slots.

Catch Zone Size

With the Catch Zone Size slider you set the range of the input pitches to be corrected to specific notes. Refer to "Setting Catch Zone Size" for details on how to use this function.

Correction Speed

Set the time is should take to adjust the pitch to the set scale. See "Setting Correction Speed" for more details.

Preserve Expression

Set how much vibrato in the input audio should be let through when you use a fast Correction Speed setting. See "About the Preserve Expression parameter" for more details.

Transpose section



Click the Transpose button to activate the Transpose section. When active, the output pitch will be transposed according the settings of the Semi and Cent parameters. The Transpose function can be used either on pitch corrected signals (with the Pitch Adjust section active) or on non pitch corrected signals (with the Pitch Adjust section deactivated).

Semi and Cent

Set desired transposition value, in semitones and cents, with the spin controls. The range is ±12 semitones and ±50 cents.

See "Using pitch shifting (Transpose)" for more details.

Formant section



Click the Formant button to activate the formant control function. When active, the formants of the input signal will be preserved and won't move with the adjusted output pitches.

! **The Formant section settings have effect on the Pitch Adjust section and on the Transpose section, if they are active. However, the Formant section can also be used stand-alone to only displace the formants of the input signal.**

Shift

Use the Shift knob to displace the formants within an range of ±1 octave.

See "What are formants?" and "Using the Formant function" for more details.

The Output Mixer section



The Pitch Adjust and Voice Synth parameters control the output from the Neptune in the following way:

Pitched Signal

This slider controls the total output volume of the Pitch Adjust, Transpose and Formant sections routed to the Left and Right outputs (see "Audio Out").

Voice Synth

This slider controls the output volume of the Voice Synth routed either to the main Left and Right outputs or to the separate Voice Synth Left and Right outputs if these are connected (see "Voice Synth Out").

! **Note that the Voice Synth function must be activated, otherwise this control will have no effect - see "Using the Voice Synth".**

Connections

Flipping the Neptune around reveals an array of connection possibilities. Some of these are CV (control signal) related and some are audio signal related.



Sequencer Control

Note

The Note input allows you to control the pitch of either the Pitch Adjust or Voice Synth (depending on what is currently selected in the MIDI Input section - see "MIDI Input"). The pitch could be controlled from a Note CV output of a Matrix or an RPG-8, for example.

! **A Gate signal on the Gate input (see below) must also be present for the Note Input to work.**

Gate

The Gate input should be used in combination with the Note modulation input (see above). As soon as a gate signal is present, any note modulation on the Note input will be activated. The Gate could be controlled from a Gate CV output of a Matrix or an RPG-8, for example.

CV In

These control voltage (CV) inputs can be used for modulating various Neptune parameters from other devices. The inputs control the following parameters:

Bend

The Pitch Bend input allows you to control pitch bend of either the Pitch Adjust or Voice Synth (depending on what is currently selected in the MIDI Input section - see "MIDI Input"). The pitch bend could be controlled from a Pitch Bend CV output of an RPG-8, for example.

Vibrato

The Mod Wheel input allows you to control the Vibrato amount of either the Pitch Adjust section or Voice Synth (depending on what is currently selected in the MIDI Input section - see "MIDI Input"). The Vibrato could be controlled from a Mod Wheel CV output of an RPG-8, for example.

Formant

The Formant Shift input allows you to control the Shift parameter in the Formant section from a CV source on another device. The Formant Shift parameter accepts bipolar control signals (-63 to +64).

CV Out

These control voltage (CV) outputs can be used for modulating other device parameters from Neptune:

Pitch

The Pitch output allows you to control the pitch of other devices from the Pitch Adjust section, either directly or via the Transpose section (depending on if Transpose is active or not). The Pitch CV corresponds to the pitch of the pitch adjusted and transposed signal. The Pitch output could be connected to the OSC Pitch CV input of another synth device, for example.

Amplitude

Neptune features an internal envelope follower. The Amplitude output sends out a control signal from this envelope follower based on the audio input level to Neptune. The Amplitude output could be connected to the Master Volume or Level CV input of a synth device, for example.

Audio In

Route your audio input signal(s) to the Left (and Right) audio input(s) to the right on the rear panel.

! **If you want to use Neptune in mono, connect only to the Left input.**

Voice Synth Out

Connect to these outputs if you want to break out the audio from the Voice Synth as separate signals and, for example, route to a separate Mix Channel for separate processing.

! **When these are connected, the Voice Synth audio will be removed from the main Audio Out outputs.**

Audio Out

The Left and Right outputs are the main stereo outputs of Neptune. Here, the audio from the Pitch Adjust, Transpose and Formant sections are routed. If the Voice Synth Sep Output (see "Voice Synth Out") are not connected, the Voice Synth audio output are also routed to the main outputs.

! **If you want to use Neptune in mono, connect only to the Left output.**



Chapter 38
Softube Amps

Introduction



The Softube Amp at the top and the Softube Bass Amp at the bottom

The Softube Amp and Softube Bass Amp are amplifier and speaker cabinet simulators based on the renowned modelling algorithms developed by Softube. The two devices feature accurate simulations of some of the most coveted vintage amplifier and cabinet models, that you can freely mix and match. Setting up a great amp tone for your guitar or bass is easy, regardless of what type of sound you are looking for. In this chapter the two devices will be described together using the collective name “Softube Amps”. Where parameters differ between the two devices this will be duly noted. In Reason Rack Plugin the Softube Amp and Softube Bass Amp Rack Extension devices can be found on the Effects palette.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

Basic usage

The Softube Amps can be used to process any signal but are best used “live”, i.e. as insert effects on an audio track that a live instrument is connected to, so that you can monitor the effect as you play and record. When recording electric guitar or bass, the basic tonal character of the amp/cabinet very much affects how the instrument responds, which in turn affects your playing. The Softube Amps allow you to set up a good basic tone with a minimum of fuss so you can start recording straight away!

Front panel

The Softube Amps front panel can be divided into two main sections; the display area with Patch/Amp/Cabinet selectors at the top, and a set of standard amplifier tone controls below. The Amp parameters (tone controls) are described in “Amp panel controls”.

Common effect device parameters

While the specific parameters for the Softube Amps are described later in this chapter, some features and procedures are common to all effect devices. Please, refer to “Common effect device features” for information about the Input meter, the Bypass/On/Off switch and Signal Flow Graphs on the Softube Amps.

Using the Softube Amps

Loading and saving patches



Loading and saving patches in the Softube Amps is done in the same way as with any other Reason device - see "Sounds, Patches and the Browser" for more details.

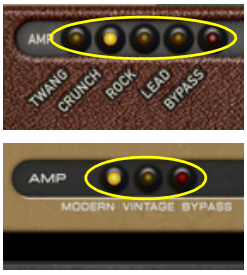
Selecting Amp and Cabinet model

There are six basic Amp models to choose from; four in the Softube Amp and two in the Softube Bass Amp. There are also four Cabinet + microphone combination models in the Softube Amp and three Cabinet + microphone combination models in the Softube Bass Amp.

You can use any Amp model with any Cabinet + microphone model. For general model descriptions see "About the Amp and Cabinet models".

Selecting Amp model

- **To select Amp model, click the desired Amp button:**



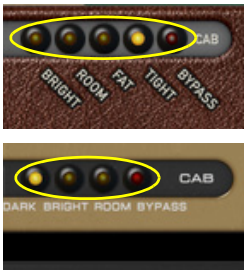
Amp selectors on the Softube Amp and Softube Bass Amp respectively.

- **To bypass the Amp section, click the Bypass button.**

Selecting Cabinet model

You can use any Cabinet with any selected Amp model.

- **To switch Cabinet model, click the desired Cab button:**



Cabinet selectors on the Softube Amp and Softube Bass Amp respectively.

- **To bypass the Cabinet section, click the Bypass button.**

About the Amp and Cabinet models

Softube Amp

The following Amp models are available in the Softube Amp:

Amp Model	Amp modeled on
Twang	A silvery American classic, great for clean and slightly overdriven guitars. Popular for country, blues and more.
Crunch	British 60's classic that keeps its chiming, present character whether you play it clean or drive it so hard it's on the verge of breaking down.
Rock	A yummy, warm, fat, tube distortion. Just like Mama used to make it.
Lead	The boogie man has come in all his high gain glory! Screaming leads and power chords that sustain for days.

The following Cabinet models are available in the Softube Amp:

Cabinet Model	Cabinet + microphone modeled on
Bright	A 2x12 cabinet with deep lows and heavenly highs. The oh-so-familiar dynamic mic is placed on axis but still lets a hint of room into the sound.
Room	A more midrange present 2x12, with the small diaphragm dynamic mic placed off axis and a wide stereo room added.
Fat	Fat, solid 4x12 classic. Captured with a German condenser microphone on axis for that full bodied sound.
Tight	Slightly midrange scooped 4x12 cabinet. The large diaphragm dynamic microphone gives an in your face presence.

Softube Bass Amp

The following Amp models are available in the Softube Bass Amp:

Amp Model	Amp modeled on
Modern	A contemporary version of a classic tube design from one of the most popular bass amplifier brands. Modern has all the clarity and punch you'd expect from a modern amplifier, but still with the pleasant three-dimensionality that well designed tube amplifiers offer.
Vintage	Got grit? Vintage gives you a dark, dirty, steamy bass sound that adds power and attitude, and helps making the bass instrument cut through the mix.

The following Cabinet models are available in the Softube Bass Amp:

Cabinet Model	Cabinet + microphone modeled on
Dark	A classic 8x10 cabinet with a snappy response across the frequency spectrum. Captured with a German large membrane dynamic microphone, straight on axis for a clear sound and a rock solid low end.
Bright	A British 4x12 bass cabinet with great midrange presence. The dynamic microphone is placed off axis for a slightly roomy sound.
Room	A 4x10 cabinet, captured with a German classic large membrane condenser mic at a distance. Roomy with a nice depth and width that caters for a very natural sound in the mix.

Amp panel controls

Softube Amp



The panel controls are used for tweaking the sound. The following controls can be found in the Softube Amp:

Control	Description
Gate knob	Here you can set a gate threshold for the Softube Amp. Signals below the set threshold are not let through.
Boost switch	This boosts the input gain, for more crunch.
Gain	This controls the input gain. The higher the Gain value, the more crunch you get.
Bass/Mid/Treble controls	These are tone controls which you can use to cut or boost the bass, midrange and high frequencies, respectively.
Poweramp Gain	This is the gain control for the modeled power amp.
Volume	Controls the master volume of the Softube Amp.

Softube Bass Amp



The panel controls are used for tweaking the sound. The following controls can be found in the Softube Bass Amp:

Control	Description
Drive	This controls the input gain. The higher the Drive value, the more crunch you get.
Bass/Middle/Mid Freq/Treble controls	These are tone controls which you can use to control the bass, midrange and high frequencies, respectively. The Bass and Treble controls also have Ultra Lo/Hi buttons for emphasizing the low and hight frequency ranges even further. There is also a Mid Freq selector where you can select different Mid Frequency EQ characteristics.
Volume	Controls the master volume of the Softube Bass Amp.

Connections



- **Both Softube Amps have Left/Right inputs and outputs.**
 If you only connect in mono to the Left input, the output will still be in stereo (if you connect both outputs).
- **The Softube Amp has CV inputs for controlling Gate, Gain and Volume.**
- **The Softube Bass Amp has CV inputs for controlling Drive and Volume.**

Chapter 39
Audiomatic
Retro Transformer



Introduction



The Audiomatic Retro Transformer Rack Extension effect device was designed mainly with focus on spicing up dull mixes. The Audiomatic Retro Transformer is inspired by the Hipstamatic picture editing app - but designed for audio.

By selecting one of the 16 presets you can instantly change the character of your sound - almost like applying a "magic skin".

At moderate levels the Audiomatic Retro Transformer can breathe life into your individual tracks, as well as into your final mixes, by adding a subtle "shimmer" to your sound. Used to its extremes, it can provide hard-edge and aggressive sounds as well. You can easily control the mix between dry and processed signal with the Dry/Wet knob on the front panel.

In Reason the Audiomatic Retro Transformer Rack Extension device can be found on the Effects palette.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

Using Audiomatic Retro Transformer

The Audiomatic Retro Transformer is designed to be used mainly as an insert effect, on individual tracks as well as in the master section. Another very nice application is to use the Audiomatic Retro Transformer as an insert effect in parallel mixer channels. Then, you can sculpt the effect sound even further using e.g EQ and compressor, for example.

The front panel layout is very simple and straight-forward: the device features 16 Preset buttons, a Transform knob, an input Gain knob and a Dry/Wet knob.

Gain



→ **Set the input level Gain.**
Range: - INF to + 12 dB.

! **Use the Gain control cautiously when using Audiomatic Retro Transformer on the whole mix. It is easy to get unwanted distortion in some of the Presets, e.g. Tape, when the input signal is loud.**

← **Don't be afraid to push the gain up - especially in presets that feature built-in compression! High input Gain settings can also render very interesting distorted sonic results. You can always compensate the level by lowering the Volume, see "Volume".**

The Presets



The 16 Preset buttons are arranged in a four by four matrix. The upper row of presets contains subtle effects, the second row a little less subtle and so on all the way down to the fourth row, which contains the most far out effects.

The display shows a picture which reflects the selected effect Preset.

The following Presets are available:

Tape

This simulates the character of an analog tape recorder.

Hi-Fi

This simulates the classic Loudness compensation function, which was very popular in numerous consumer class hi-fi systems in the 70s and 80s.

Bright

This preset adds brightness to the sound, and removes bass.

Bottom

This preset tightens up and enhances the low frequencies in the sound.

Spread

This preset is a spatial effect, which spreads the stereo width of the sound and also changes the frequency characteristics.

Radio

This simulates a small transistor radio.

VHS

This simulates sound recorded with a VHS camera.

Vinyl

This simulates the background sound and noise from a somewhat scratchy vinyl record.

MP3

This simulates the sound of a poorly encoded/decoded MP3 file.

Psyche

A psychedelic sound experience. “Turn on, tune in, drop out”, sort of.

Cracked

The effect of a broken speaker with lots of distortion.

Gadget

A hollow “robotic” type of effect.

Circuit

A “circuit-bending” type of effect with bit-crushing on top.

Wash

A washing machine on sound check at an open air venue?

PVC

A pretty far out sci-fi preset.

Eerie

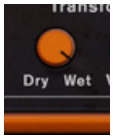
A very nice and scary waterphone effect.

Transform



- **Transform the effect with the Transform knob.**
 The knob usually controls a combination of different (hidden) parameters in each of the Presets. This makes it behave more like a “morph” control rather than an “amount” control.

Dry/Wet



- **Set the balance between the dry input signal and the transformed (effect) signal with the Dry/Wet knob.**
 - **If you want only subtle effects, turn the Dry/Wet knob more towards the Dry position. For more prominent effects, turn the Dry/Wet knob towards the Wet position.**

Volume



→ **Set the output volume with the Volume knob.**
Range: - INF to + 12 dB.

← **Use the Volume knob in combination with the "Gain" knob, to compensate for high input Gain settings.**

Connections



CV Modulation In

Transform

A bipolar CV signal patched here modulates the Transform parameter (see "Transform"). A positive signal increases the Transform parameter amount and a negative signal decreases it.

← **Patch an LFO signal here and use with the PVC Preset, for a nice sweeping sound!**

! **Note that the default Transform parameter range can not be exceeded.**

Dry-Wet

A bipolar CV signal patched here modulates the Dry/Wet front panel control (see "Dry/Wet"). Zero modulation means that the current Dry/Wet knob setting is valid.

Input L&R

→ **Patch the audio signals you want to process here.**
If your signal is in mono, connect only to the L (left) input.

Output L&R

These are the stereo audio outputs.

- **The signal routing (stereo/mono/dual mono) is Preset dependent.**

Chapter 40
Channel Dynamics
Compressor & Gate

Introduction



Channel Dynamics Compressor & Gate is the rack version of the Dynamics section in Reason's Main Mixer. The rack version is mainly intended for use in the VST3 plugin version of Reason, since it lacks the Main Mixer. However, you can of course use it wherever you like - also in the stand-alone Reason program. The Channel Dynamics Compressor & Gate device has identical specifications compared to the channel strip version, with the addition of Input Gain and Dry/Wet Mix controls.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Panel reference

Global controls

Input Gain



→ **Adjust the input gain to the Channel Dynamics device.**
Range: +/- 18.00 dB

! **The Input Gain knob controls the input signal, regardless if the Compressor and/or Gate is on or not.**

Mix



→ **Set the mix between dry and processed signal with the Mix knob.**
By setting a mix of <100% you can use the Channel Dynamics device for "parallel processing", i.e. the dry and effect signals are being output together.

The Compressor section



Compressors reduce dynamic range by evening out the difference between loud and quiet signals. This makes signal levels easier to balance, and can add punch and sustain to the sound.

The compressor/limiter in Channel Dynamics is a flexible processor which has soft-knee (a gradual, smooth onset of compression) characteristics but can be switched to peak limiting, where levels above the set threshold are instantly reduced. The compressor also features automatic make-up gain. The parameters are as follows:

On

- **Click the On button to activate the Compressor section.**

Peak

- **Activate to change the signal detection from RMS to Peak, which results in an instant attack time.**
 Peak mode is suitable for compression of sounds with fast attacks like drums.

Fast

- **Click this to make the compressor react to signals above the set Threshold level in a fixed time of 3 ms for 20 dB gain reduction.**

Ratio

- **Set the amount of gain reduction applied to the signal above the set Threshold (see below).**
 Range: 1:1 (no reduction) to Infinite:1

Threshold

- **Set the level at which onset of compression should occur.**
 Signals below the Threshold setting are unaffected, but when the level exceeds the threshold, compression kicks in. Automatic make-up gain (based on the Ratio and Threshold settings) is applied to compensate for level reduction caused by compression, to maintain a steady output level.

 Range: -52 dB to 0 dB

Release

- **Set the time it should take before the compressor lets the sound through unaffected - after the signal level drops below the set threshold.**
 Set this to short values for “pumping” compressor effects, or to longer values for a smoother change of the dynamics.

 Range: 100 ms to 1000 ms

Compressor Gain Reduction Meter

The LED meter shows the gain reduction applied by the compressor.

The Gate/Expander section



Gating or expansion will attenuate signals below a set threshold; the opposite of compression. It can be used to reduce or eliminate unwanted background noise that may be present when there is no signal to mask it. Gating is also commonly used to reduce microphone "bleeding", e.g. when recording a close-mic'ed drum kit you can use gating to silence the tom microphones when the toms aren't being played to tighten up the sound, and for special effects like "keying" (see below).

Higher expansion ratios (10:1 and above) are referred to as noise gating, where the channel is completely silenced if the level drops below the set threshold.

The Gate/expander has the following parameters:

On

→ **Click the On button to activate the Gate/Expander section.**

Exp

In Expander mode, the Range (gain reduction) is not as severe as in Gate mode, but more gradual around the Threshold. Also, the gain reduction is not attenuated by a certain dB but is scaled. The major difference compared to the Gate function is that instead of muting the signal once it drops below the Threshold (gate), the Expander still lets some signal through but at a lower level, resulting in a signal of lower volume. You could think of an Expander as an "inverted" compressor, i.e., it expands the dynamic range of the input signal.

→ **Click the Exp button to change the operating mode from Gate to Expansion.**
The Range knob (see below) then controls the expansion amount.

Hold

→ **Set the time the gate should stay fully open after the signal falls below the Threshold.**
Hold interacts with the Release parameter such that Release only starts acting after the set Hold time.
Range: 0 ms to 4000 ms

Fast

The normal attack time for the Gate is normally1.5 ms per 40 dB.

→ **Click the Fast button to lower the attack time to 100µs (microseconds) per 40 dB.**
This can be is useful for percussive material were the waveform rises steeply in a very short time.

Range

→ **Set the amount of gain reduction applied to signals below the set Threshold.**
The Range can be set from 0 dB (no reduction) to -40 dB. If the Exp button is on (see above), the Range knob controls the expansion amount.

Threshold

→ **Set the level at which the gate opens or closes.**
Signals below the Threshold setting are gated, but when the level exceeds the threshold, the gate opens.
Range: -52 dB to 0 dB

! **Note that the threshold for closing the gate is slightly lower than the threshold for opening the gate. This is to avoid undesirable gate triggering with signal levels close to the set threshold.**

Release

→ **Set the time it should take for the gate to go from open to fully closed.**
Fast release times will fade the signal abruptly once the level falls below the threshold, and longer release times will slowly fade out the signal.

Range: 100 ms to 1000 ms

Gate Gain Reduction Meter

The LED meter shows the gain reduction applied by the gating/expansion.

External Sidechain



You can use external signals to trigger the Compressor and/or the Gate/Expander. This is done by connecting an external signal to the Sidechain Inputs on the back of Channel Dynamics.



The Sidechain inputs.

When a cable is connected to the Sidechain Input(s) the "Connected" LED is lit on the front panel:



→ **Click the Sidechain button to "key" (trigger) Channel Dynamics from the external signal instead of the channel signal.**
For example, you could use a drum loop to trigger the gate for a channel playing a synth pad to create rhythmic chord effects.

Connections



CV Outputs

These two CV modulation outputs can be used for controlling other devices that feature CV modulation inputs. The modulation parameters are:

- **Compressor Gain Reduction (see "Compressor Gain Reduction Meter")**
- **Gate Gain**
 This CV signal is high when the gate is open and goes low when the gate closes.

Sidechain Input Left & Right

→ **Patch an external audio signal to be used as sidechain signal here.**
See "External Sidechain" for more information.

Input Left & Right

→ **Patch the audio signals you want to process here.**
If your input signal is in mono, connect only to the L (left) input.

Output Left & Right

These are the audio outputs.

Chapter 41
Channel EQ Equalizer

Introduction



Channel EQ Equalizer is the rack version of the EQ section in Reason's Main Mixer. The rack version is mainly intended for use in the VST3 plugin version of Reason, since it lacks the Main Mixer. However, you can of course use it wherever you like - also in the stand-alone Reason program. The Channel EQ device has identical specifications compared to the channel strip version.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Panel reference

Global controls



Gain

→ **Adjust the Channel EQ gain with this knob.**
This is useful for level-compensating when doing drastic boosts or cuts.
Range: +/- 18.00 dB

The signal level is shown in the meter.

The Filter section



The Filter section contains a 12 dB/octave low pass filter and an 18 dB/octave high pass filter.

The parameters are as follows:

HPF On

→ **Click the On button to activate the high pass filter.**

HPF Freq

The High Pass filter (HPF) removes low frequencies from the signal, resulting in a thinner sound. The HP filter slope has a 18 dB/Octave roll-off.

→ **Set the desired cutoff frequency for the high pass filter.**
Range: 20 Hz - 4 kHz

LPF On

→ **Click the On button to activate the low pass filter.**

LPF Freq

The Low Pass filter (LPF) removes high frequencies from the signal, making the sound less bright. The LP filter has 12 dB/Octave roll-off curve.

→ **Set the desired cutoff frequency for the low pass filter.**
Range: 100 Hz - 20 kHz.

The Equalizer section



The Equalizer section features a four-band EQ with parametric midrange controls and high and low frequency shelving bands. The EQ can be switched between two operating modes, each with slightly different curve characteristics.

The Equalizer has the following parameters:

LF Gain

The LF section provides low frequency shelving equalization. All frequencies below the set LF Frequency will be cut or boosted by the set LF Gain amount.

→ **Set the LF Gain/Attenuation amount.**
Range: +/- 20 dB

LF Frequency

→ **Set the LF Frequency.**
Range: 40 Hz - 600 Hz

LF Bell

→ **Click the LF Bell button to switch the LF EQ to peaking characteristics.**
This means it works like a regular parametric EQ band, cutting or boosting the signal around the set LF Frequency. Bell mode has a fixed bandwidth or "Q" value.

LMF Gain

The low medium frequency EQ is fully parametric.

→ **Set the LMF Gain/Attenuation amount.**
Range: +/- 20 dB

LMF Frequency

→ **Set the LMF (center) Frequency.**
Range: 200 Hz - 2 kHz

LMF Q

The “Q” parameter adjusts the bandwidth around the set center LMF Frequency. The higher the Q value, the narrower the affected frequency range - except in “E” mode (see “E Mode”).

→ **Set the Q value for the LMF EQ.**
Range: 0.70 - 2.50

E Mode

→ **Click to switch to E Mode.**
When the E Mode button is activated the EQ will have slightly different curve characteristics. In normal mode (E button deactivated), the Gain setting will also affect the bandwidth (Q value) for the LMF and HMF EQs. The higher the Gain, the narrower the bandwidth and vice versa.

With E Mode activated, the bandwidth is constant at all Gain settings.

HMF Gain

The high medium frequency EQ is fully parametric.

→ **Set the LMF Gain/Attenuation amount.**
Range: +/- 20 dB

HMF Frequency

→ **Set the HMF (center) Frequency.**
Range: 600 Hz - 7 kHz

HMF Q

The “Q” parameter adjusts the bandwidth around the set center HMF Frequency. The higher the Q value, the narrower the affected frequency range - except in “E” mode (see “E Mode”).

→ **Set the Q value for the LMF EQ.**
Range: 0.70 - 2.50

HF Gain

The HF section provides high frequency shelving equalization. Frequencies above the set corner Frequency will be cut or boosted by the set HF Gain amount.

- **Set the HF Gain/Attenuation amount.**
 Range: +/- 20 dB

HF Frequency

- **Set the HF Frequency.**
 Range: 1.5 kHz - 22 kHz

HF Bell

- **Click the HF Bell button to switch the HF EQ to peaking characteristics.**
 This means it works like a regular parametric EQ band, cutting or boosting the signal around the set HF Frequency. Bell mode has a fixed bandwidth or "Q" value.

Connections



CV Inputs

These six CV modulation inputs, with associated trim pots, can be used for controlling the following parameters from CV modulation sources:

- **HPF Frequency (see "HPF Freq")**
- **LPF Frequency (see "LPF Freq")**
- **HMF Gain (see "HMF Gain")**
- **HMF Frequency (see "HMF Frequency")**
- **LMF Gain (see "LMF Gain")**
- **LMF Frequency (see "LMF Frequency")**

Input Left & Right

→ **Patch the audio signals you want to process here.**
If your input signal is in mono, connect only to the L (left) input.

Output Left & Right

These are the audio outputs.

Chapter 42
Master Bus Compressor

Introduction



The Master Bus Compressor is the rack version of the Master Bus Compressor in Reason's Main Mixer. The rack version is mainly intended for use in the VST3 plugin version of Reason, since it lacks the Main Mixer. However, you can of course use it wherever you like - also in the stand-alone Reason program.

The Master Compressor is perfect for providing the final "fairy dust" to your mix. It can add punch and cohesion, and generally make the mix sound bigger and more powerful. It's perfect for use on a drum channel or in a mixer bus. The compressor is very straightforward in operation and features make-up gain as well as program-adaptive Release.

The Master Bus Compressor device has identical specifications compared to the Master Section version, with the addition of Input Gain and Dry/Wet Mix controls.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Panel reference

Global controls



Mix

→ **Set the mix between dry and processed signal with the Mix knob.**
By setting a mix <100% you can use the Master Bus Compressor for parallel processing, i.e. the dry and effect signals are being output together.

Compression controls



Compressors reduce the dynamic range by evening out the difference between loud and quiet signals. This makes signal levels easier to balance, and can add punch and sustain to the sound, as well as “glue” together a final mix. The parameters are as follows:

Input Gain

- **Adjust the input gain to the Master Bus Compressor.**
 Range: +/- 18.00 dB

Threshold

- **Set the level at which onset of compression should occur.**
 The lower the Threshold, the more compression is applied.
 Range: -30 dB to 0 dB

Ratio

Ratio specifies the amount of gain reduction applied to signal levels above the set Threshold. A 2:1 compression ratio effectively means that a signal level 2dB above threshold will have a signal gain of 1dB.

- **Set the amount of gain reduction applied to the signal above the set Threshold.**
 Range: 2:1, 4:1 and 10:1

Attack

- **Set the time it should take before the compressor should react to signals above the set Threshold.**
 Range: 0.1 ms, 0.3 ms, 1 ms, 3 ms, 10 ms and 30 ms.

Release

- **Set the time it should take before the compressor lets the sound through unaffected - after the signal level drops below the set Threshold.**
 If set to “Auto”, the Release time will be program adaptive, so the Release time is automatically increased following long peaks and decreased following short peaks.
 Range: 0.1 s, 0.3 s, 0.6 s, 1.2 s and “Auto”.

Make Up

Make-Up gain compensates for level reduction caused by compression and helps maintain a steady output level.

- **Adjust the output gain from the Master Bus Compressor device.**
 Range: - 5 dB to + 15.00 dB

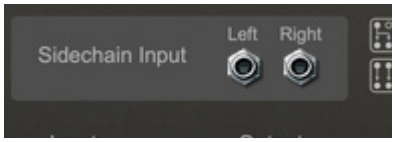
Compressor Gain Reduction Meter

The meter shows the gain reduction amount in dB applied by the compressor.

External Sidechain



You can use external signals to trigger the Master Bus Compressor. This is done by connecting an external signal to the Sidechain Inputs on the back of the Master Bus Compressor.



The Sidechain inputs.

When a cable is connected to the Sidechain Input(s) the "Connected" LED is lit on the front panel:



- **Click the Sidechain button to "key" (trigger) Master Bus Compressor from the external signal instead of the channel signal.**
 For example, you could use a kick drum pattern to trigger the sidechain function, to achieve a rhythmic pumping effect.

Connections



Comp Gain Reduction

This CV output sends out the gain reduction as a CV signal (see “Compressor Gain Reduction Meter”).

Sidechain Input Left & Right

- **Patch an external audio signal to be used as sidechain signal here.**
 See “External Sidechain” for more information.

Input Left & Right

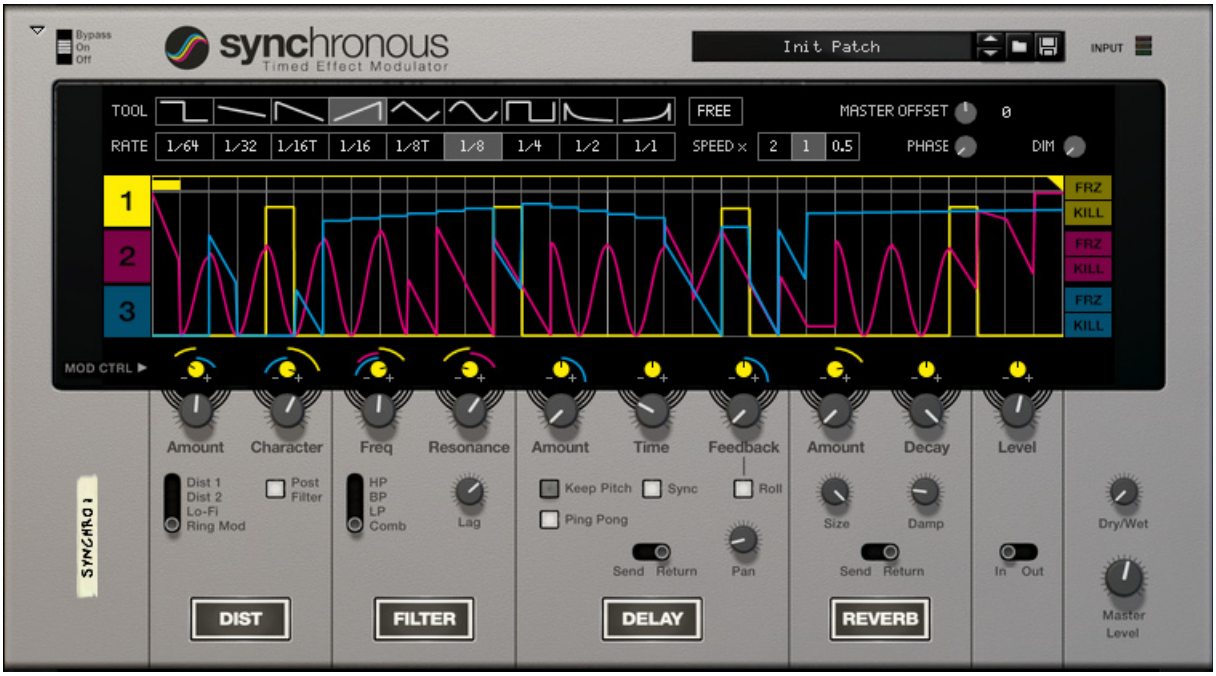
- **Patch the audio signals you want to process here.**
 If your input signal is in mono, connect only to the L (left) input.

Output Left & Right

These are the audio outputs.

Chapter 43
Synchronous
Timed Effect Modulator

Introduction



The Synchronous Timed Effect Modulator device is a very flexible multi effect “loop” device with freely designable effect parameter modulation curves. Synchronous features built-in Distortion, Filter, Delay and Reverb effects that can be modulated simultaneously by up to three modulation curves.

By drawing your own unique modulation curves in the display and assigning these curves to the desired effect parameters, you get a very flexible system for repeatedly modulating the effects parameters - in perfect sync with the main sequencer.

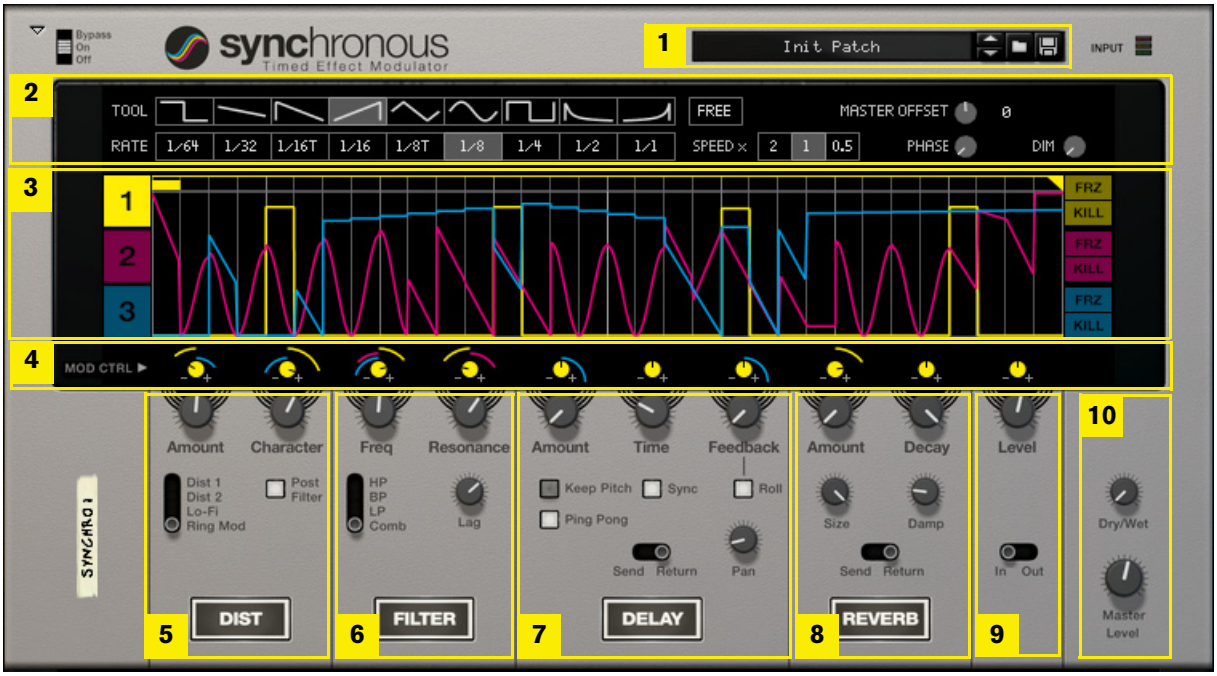
The standard loop length is 2 bars. However, if you run Synchronous in half speed, you will get a maximum loop length of 4 bars.

Synchronous is designed to be used mainly as an insert effect, on individual tracks, or as a “loop mangling” effect hooked up to the outputs of a Dr Octo Rex or Redrum device, for example.

! Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.

Panel overview

Below is a description of the different sections in Synchronous:



Synchronous panel sections

- **1. Patch selector (for browsing, loading and saving patches).**
- **2. Curve design tools, rate, speed, phase, master offset and curve dimming controls.**
- **3. Modulation curves display with selection, freeze, kill and loop length controls for each curve.**
 The display is where you draw your effect modulation curves.
- **4. Modulation control section.**
 Here is where you set how much each modulation curve should affect each effect.
- **5. Dist section.**
 Features two distortion effects, ring modulation, and a lo-fi effect.
- **6. Filter section.**
 Features lowpass, bandpass, highpass and comb filter types.
- **7. Delay section.**
 Features regular delay, ping-pong delay and roll (freezed delay) effects.
- **8. Reverb section.**
 Features a stereo reverb.
- **9. Level modulation section.**
- **10. Master dry/wet and level controls.**

Using Synchronous

Loading and saving patches



Loading and saving patches is done in the same way as with any other internal Reason device, see "Loading patches" and "Saving patches" for details.

Drawing and assigning modulation curves - a tutorial

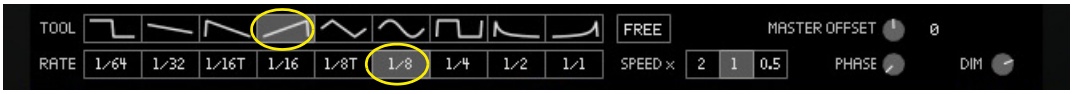
To describe the basic principles with modulation curves, let's have a look at an example of how you can design a modulation curve and then have it modulate a couple of effect parameters:

1. **Create a Dr Octo Rex device.**
 A Dr Octo Rex device is created and the default Rex loops are loaded in the device.
2. **Select the Dr Octo Rex device in the rack and create a Synchronous device.**
 The audio outputs of the Dr Octo Rex device are automatically connected to the audio inputs of the Synchronous device. The audio outputs of the Synchronous device are also automatically routed to the Dr Octo Rex's Mix Channel.
3. **Select the Synchronous device in the rack and select "Reset Device" from the context menu or Edit menu.**
 All parameters in the Synchronous device are now reset to their default values. The yellow modulation curve is automatically selected and reset to a straight line down in the display.
4. **Click Play on the sequencer Transport Panel in Reason.**
 The default Rex loop in the Dr Octo Rex device starts to play back and the Position Indicator at the top of the Synchronous display starts to advance in sync with the sequencer.

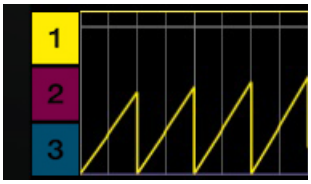


The Position Indicator advances in sync with the sequencer tempo.

5. **Stop the sequencer.**
6. **Click the Positive Sawtooth Tool button and then the 1/8 Rate button:**



7. **Place the mouse pointer to the far left on the display, in the curve area.**
8. **Click-hold and then drag the mouse pointer to the right on the display.**
9. **Release the mouse button when you have reached the second thicker vertical grid line.**
 A sawtooth wave is now visible in the display. The rate of the sawtooth wave is 1/8th of a bar, since you selected this when you clicked the 1/8 button. The amplitude of the sawtooth wave is increased or decreased according to your vertical drawing direction.



10.Now, click the Stepped Line Tool button and the Free button:



The Rate buttons have now disappeared from the panel, since they are of no importance when you use the Line tools.

11.Click-hold the mouse pointer at the second thick line and then draw to the right throughout the rest of the display area.
Alter the amplitude (y-axis positions) as you draw.

12.Release the mouse button when you have reached the far right of the display.
Now, you have half a bar of a 1/8th sawtooth waves and one and a half bars of a stepped "free hand" curve:



Now, let's assign the yellow modulation curve to the effect parameters to be modulated. In this example we will assign the curve to the Freq and Resonance parameters of the Filter section:

1. **Click the Filter button to switch on the Filter section:**



2. **Turn the Freq knob to a low value and the Resonance knob to a high value:**



3. **Raise the yellow Freq Modulation knob (above the Freq knob) to a positive value (past its 12 o'clock position). Then, set the Resonance Modulation knob (above the Resonance knob) to a negative value.**
Yellow modulation amount semicircles appear around the Freq Modulation knob and around the Resonance Modulation knob, to indicate the modulation amounts and their "directions" (positive or negative modulation):



Setting the modulation amounts.

4. **Start the sequencer.**
Congratulations, you have now created your first looped Filter modulation in Synchronous!

Editing modulation curves - a tutorial

Basically, there is one way you can edit an existing Modulation Curve: by redrawing the part of the curve you want to change. The examples below describe two basic use cases.

Changing an existing curve's amplitude

If you want to adjust the modulation curve's amplitude anywhere throughout the modulation loop, proceed as follows:

1. **Click the Curve Select button for the modulation curve you want to edit.**
 The curve is highlighted in the display.
2. **Click the Tool button which corresponds to the existing curve in the display.**
 If the curve is a waveform, make sure you also click the correct Rate button.
3. **Now, you have two options:**

→ **To change the amplitude in a linear fashion (with a constantly increasing or decreasing amplitude), make sure the Free button is deselected (off). Place the mouse pointer where you want the change to begin and then draw to the right to where you want the change to end.**
The amplitude starts at the level where you placed the mouse pointer and ends at the level where you released the mouse button. The level change between the start and end point is linear.

→ **To change the amplitude individually between each vertical grid line, make sure the Free button is on. Place the mouse pointer where you want the changes to begin and then draw to the right to where you want the changes to end.**
The amplitude starts at the level where you placed the mouse pointer, then follows your drawing directions, and ends at the level where you released the mouse button. The levels changes are now individual between each vertical grid line.

! **Note that the shortest interval you can replace (or add) curves in is in 1/16th note "steps" of a bar, i.e. in between two vertical grid lines in the display (in Speed x 1 mode). The mouse pointer automatically "snaps" to the closest leftmost vertical grid line when you draw.**

← **If you are not satisfied with your amplitude changes, just use the standard [Ctrl](Win)/[Cmd](Mac)+[Z] command to undo the operation.**

Replacing a part of a curve with another one

To replace the current modulation curve with a different curve anywhere in the modulation loop, use the same principle as described above in "Changing an existing curve's amplitude". The only exception here is that you now freely choose the new curve shape, regardless of the existing curve shape. You can replace an existing curve in as many places throughout the loop as you like.

! **Note that the shortest interval you can replace (or add) curves in is in 1/16th note "steps" of a bar, i.e. in between two vertical grid lines in the display (in Speed x 1 mode). The mouse pointer automatically "snaps" to the closest leftmost vertical grid line when you draw.**

← **If you accidentally overwrite a curve part that you want to keep, just use the standard Undo command to undo the operation.**

Panel reference

The display section



Tool buttons and the Free button



- **Click the Stepped Line button when you want to draw a curve that has fixed levels between each vertical grid line in the display:**

 The result when you draw will be a jagged curve with vertical steps.

- **Click the Linear Line button when you want to draw straight lines between the vertical grid lines in the display:**

- **Click any of the Waveform buttons when you want to draw (repetitive) waveforms in the display:**

 In these situations, the Rate buttons define the waveform cycle lengths, see “Rate buttons”.

- **Click the Free button in combination with a Line button to be able to freely define the amplitude between each of the vertical grid lines in the display:**

- **Click the Free button in combination with a Waveform button to be able to freely define the waveform’s amplitude between each of the vertical grid lines in the display:**

Rate buttons



- **Use the Rate buttons to define the waveform cycle length when you draw waveforms in the display.**
 You can alter between different time rates throughout the loop by clicking another Rate button and continue drawing from where you stopped.

! **When you have selected a Line tool, the Rate buttons automatically become invisible.**

Speed control



→ **Use the Speed x control to set the playback speed of the currently selected modulation curve.**
The Speed can be set to 2, 1 or 0.5 times the main sequencer tempo, and are set individually for each of the three modulation curves. The speeds are indicated by the Position Indicators' advancement when the main sequencer is running.

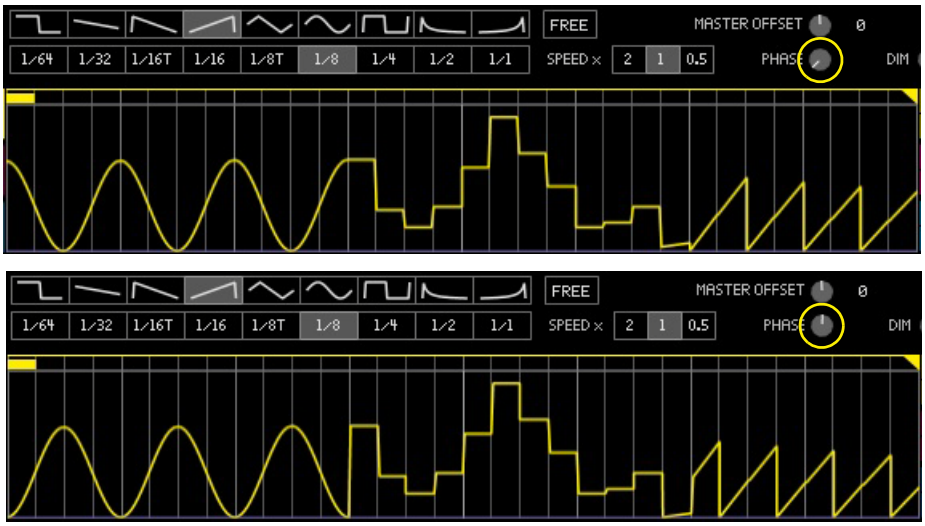
- **In Speed x 1, the display covers 2 bars.**
The vertical grid lines in the display represent 1/16th note.
- **In Speed x 0.5, the display covers 4 bars.**
The vertical grid lines in the display now represent 1/8th note.
- **In Speed x 2, the display covers 1 bar.**
The vertical grid lines in the display now represent 1/32th note.

Phase knob



→ **Set the phase of the currently selected modulation curve.**
Range: 0-360 degrees.

The phase is changed individually for each curve type (waveform) used in the loop. The picture below shows the same modulation curve with a Phase setting of 0 and ~180 degrees, respectively:



The same modulation curve at 0 degrees (top) and 180 degrees (bottom) Phase values.

As you can see in the picture above the first part of the curve, which is a sinewave with a 1/4th note rate is phase shifted 180 degrees. The second part, which is a stepped line, is not changed at all. The third part of the curve, which is a sawtooth wave with a 1/8th note rate is also phase shifted 180 degrees. However, since the third part has a 1/8th note rate, the actual "distance" it has been moved is half as long as the 1/4th note rate sinewave in the first part of the curve. This is because all waveform curves are phase shifted individually throughout the modulation curve.

! **Stepped Line and Linear Line curves are not affected by the Phase parameter - they stay fixed regardless of the Phase value.**

← **If you want to "time shift" the entire modulation curves, including the Line curves, use the Master Offset function described below.**

Master Offset knob



→ **Set the common offset for all three modulation curves.**
This determines where in the modulation loop the playback should begin. Negative Master Offset values moves the playback start position backwards, and vice versa.

! **The Master Offset is indicated only by the Position Indicators - i.e. the curves are not displaced in the display.**

Range: +/- 32 16th notes (in Speed x 1 mode).

Range: +/- 32 8th notes (in Speed x 0.5 mode).

Range: +/- 32 32th notes (in Speed x 2 mode).

Dim knob



→ **Turn the Dim knob to adjust the dimming amount of the currently unselected modulation curves.**
At zero, all modulation curves are equally bright all the time.
At maximum level, only the selected modulation curve is visible in the display.

! **Note that the Dim value is also saved with the patch.**

Curve Select buttons

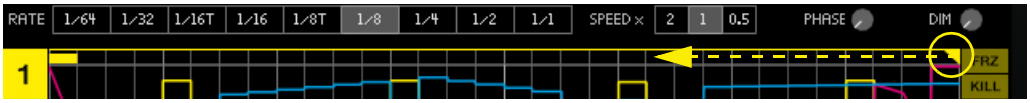


→ **Click one of these buttons to select the modulation curve you want to edit.**

Setting the loop lengths

The three modulation curves can have their own individual loop lengths. The loop length for a modulation curve can be adjusted, simply by repositioning its loop locator:

1. **Click the Curve Select button for the modulation curve you want to edit.**
2. **Click-hold and drag the colored triangular loop locator horizontally to the new desired position.**

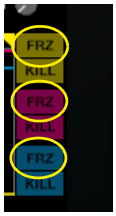


Setting the loop length for the yellow modulation curve.

3. **Release the mouse button to set the new loop length.**
The loop locator will snap to the closest leftmost vertical grid line in the display. In Normal speed, this means it will snap to a 1/16th note grid.

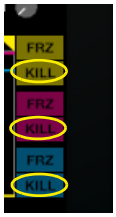
← **By setting different loop Lengths for the three modulation curves, you can get very animated variations in the modulations over longer periods of time.**

FRZ (freeze) buttons



1. **Click a FRZ (freeze) button to stop the playback of the corresponding modulation curve and freeze its current modulation value.**
2. **Click again to deactivate the freeze function.**
 The playback automatically continues at the position in the loop where it should have been if the freeze function had not been activated.

Kill buttons



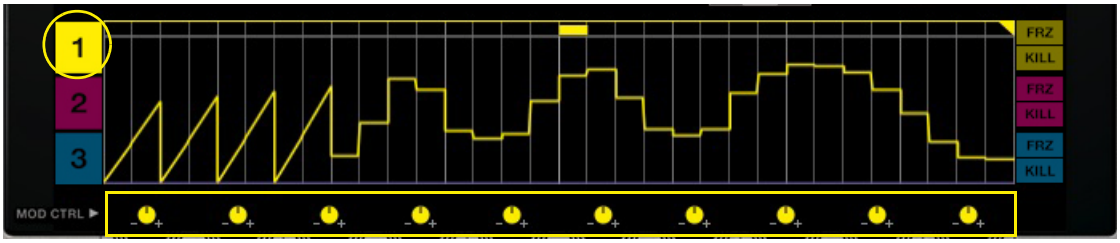
1. **Click a Kill button to deactivate (mute) the corresponding modulation curve.**
 The modulation curve is preserved but does not modulate any parameter as long as the Kill function is active. Also, the killed curve becomes invisible in the display.
2. **Click again to deactivate the Kill function.**
 The modulation curve is activated and automatically continues with the modulation at the current position in the loop.

Modulation controls

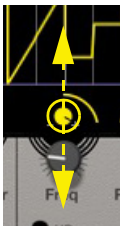


The Modulation Control knobs are used for setting a modulation amount (positive or negative) for the effect parameter right below each Modulation Control knob. To set up a modulation, proceed as follows:

1. **Select the modulation curve you want to use for the modulation by clicking its Curve Activate button.**
 All assignable Modulation Control knobs are automatically colored according to the selected modulation curve:



2. **Set the modulation amount for the corresponding effect parameter by turning the desired Modulation Control knob.**



Setting the modulation amount.

A colored semicircle appears around the Modulation Control knob. The semicircle shows how much the selected modulation curve affects the corresponding effect parameter (below each Modulation Control knob) - and in which direction (positive or negative).

→ **If you want the selected modulation curve to modulate other effect parameters, adjust the Modulation Control knobs above the other desired effect parameters.**

3. **If you want other modulation curves to modulate the same - or other - effect parameters, click the desired Curve Activate button and repeat the modulation assignment procedure from Step 1.**
All Modulation Control knobs can be modulated by all three modulation curves at the same time if you like. The modulation amounts and directions can be set independently for each modulation curve. The picture below shows a Modulation Control knob which is affected differently by all three modulation curves:



Different modulation amounts set for each of the three modulation curves.

Dist section



Amount knob

→ **Set the distortion amount.**
At 0, the signal is left unaffected.

Character knob

→ **Set the frequency content of the distorted signal.**
The effect varies depending on the selected distortion type (see below).

Distortion type selector

- **Select distortion type by clicking the corresponding LED button.**
 In Ring Mod mode, the Character knob controls the ring modulator frequency.
 In Lo-Fi mode, the Character knob controls the sample rate. There is no anti-aliasing filter in this algorithm, so there will be a lot of nice lo-fi character to the sound.

Post Filter button

- **Click the Post Filter button to route the Dist section after the Filter section (instead of before, which is the standard routing).**
 The routing order can have a big impact on the frequency content in the sound. Since the Dist section adds frequencies to the sound, you might want to preserve these by routing the Dist section after (Post) the Filter section.

Dist On/Off button

- **Click the Dist button to switch on/off the Dist effect section.**
 If switched off, the signal is bypassed unaffected.

Filter



Freq knob

- **Set the cutoff frequency (for the HP, LP and Comb filter types) or center frequency (for the BP filter type).**

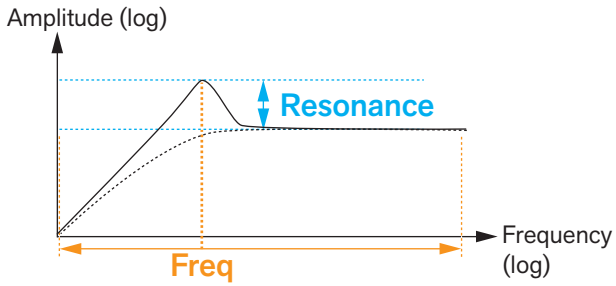
Resonance knob

- **Set the resonance amount of the filtered signal.**

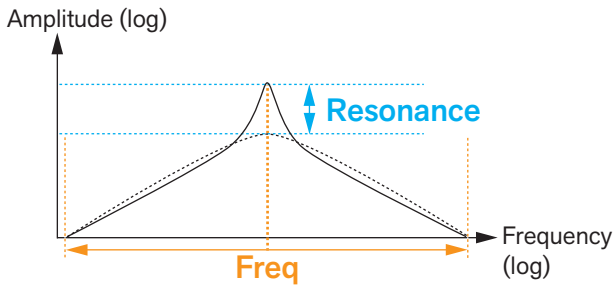
Filter Type switch

→ **Click to select the desired filter type.**
The filter types are:

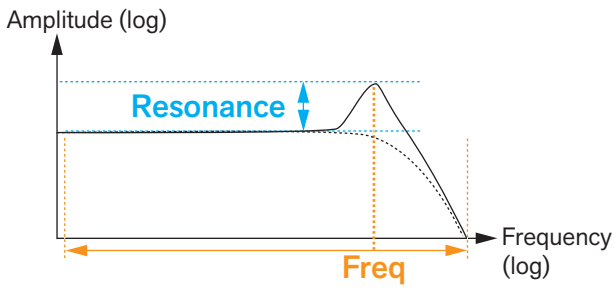
- **12 dB highpass (HP):**



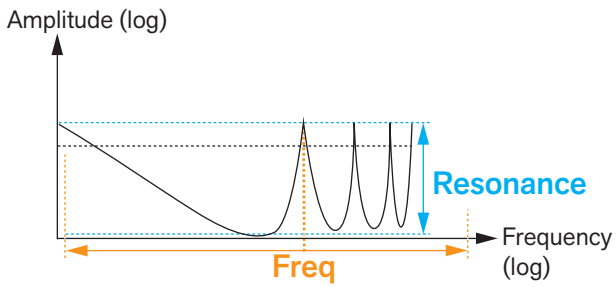
- **6 dB bandpass (BP):**



- **24 dB lowpass (LP):**



- **Comb filter with positive feedback:**



Lag knob

→ **Increase the Lag to get smoother frequency variations when the Freq parameter is modulated.**
This works like a sort of "portamento" for the Freq parameter, i.e. the Freq parameter will move more slowly when modulated if the Lag value is high.

Filter On/Off button

→ **Click to switch on/off the Filter effect section.**
If switched off, the signal is bypassed from the Filter section.

Delay



Amount knob

→ **Set the amount of the delay signal.**
At 0, the signal is left completely dry.

Time knob

→ **Set the time between the delay repetitions.**
If the "Sync button" is on, the Time values can be stepped between time divisions (e.g. 1/1, 1/2, 1/4, 1/8 etc.) relative to the main sequencer tempo.

Feedback knob

→ **Set the feedback amount, i.e. the amount of repetitions, of the delayed signal.**

Keep Pitch button

When you manually change the Rate during recording or playback, you will notice that the pitch of the delayed signal also changes. If this effect is undesirable, you can enable Keep Pitch, which will ensure that the pitch remains fixed regardless of changes in Rate.

Ping Pong button

With Ping Pong enabled, the stereo position of each delay repeat will alternate between left and right. The "Pan knob" determines the stereo width as well as the position of the initial repeat. When the Pan knob is set to full Left, the first delay bounce will be panned hard Left, the second will be panned hard Right, and so on. When the knob is set to full Right, the order is reversed (R > L > R etc).

Sync button

→ **Click the Sync button to sync the delay times to the main sequencer tempo.**
Set the time divisions (e.g. 1/1, 1/2, 1/4, 1/8 etc.) with the "Time knob".

Roll button

The Roll function works like a "freezed" delay, perfect for stutter, repeat and glitch effects. When the Roll button is on, and you turn up the Feedback knob, the input signal to the Delay section is gradually suppressed, while the feedback is automatically raised internally. In Roll mode, the Amount knob controls the level of the delay signal when the Feedback parameter is turned up. When the Feedback is set to zero, the Amount value is disregarded.

1. **Click the Roll button to switch from the regular delay settings to Roll mode.**
2. **Turn the Feedback knob to set the mix between the rolled delay signal and the dry signal.**

→ **To get an instant stutter/repeat effect, turn the Feedback knob up quickly from zero at the point where you want the stutter effect. Then, quickly turn the Feedback knob back to zero again when you want the effect to disappear.**

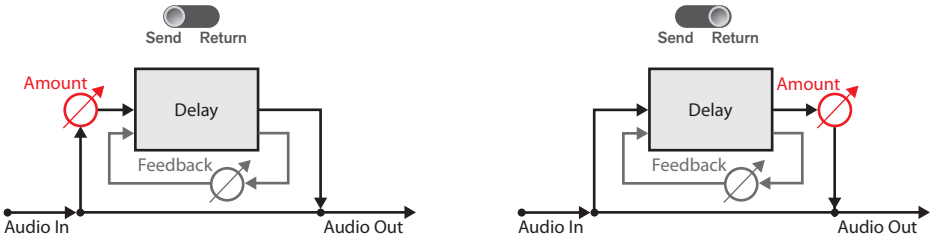
► **Experiment by changing the Time parameter to get different "stutter" times.**

Pan knob

→ **Use the Pan knob to set the stereo position of the delay repetitions. In Ping Pong mode, the Pan parameter defines where the initial delay bounce should be placed in the stereo panorama, see "Ping Pong button".**

Send/Return switch

→ **Select whether you want the Amount knob to control the Send level or the Return level of the delay effect.**
The picture below shows the different configurations schematically:



The schematic placement of the Amount knob in Send and Return mode, respectively.

Delay On/Off button

→ **Click to switch on/off the Delay effect section.**
If switched off, the signal is bypassed from the Delay section.

Reverb



Amount knob

→ **Adjust the balance between the unprocessed and the reverberated audio signal.**

Decay knob

→ **Set the decay time of the reverberated signal.**

Size knob

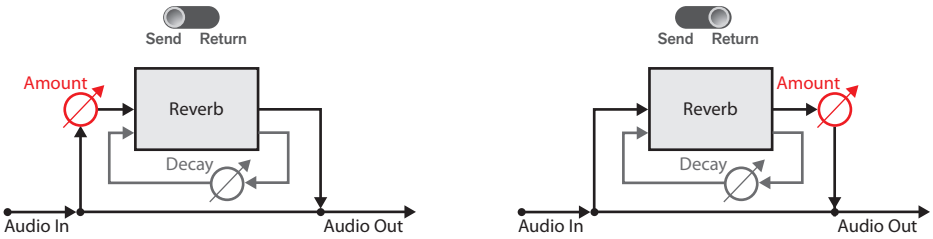
→ **Set the emulated room size, from small room to large hall, with the Size knob.**
Lowering this parameter results in a closer and gradually more "canned" sound. Raising the parameter results in a more spacey sound, with longer pre-delay.

Damp knob

→ **Set the high-frequency damping amount of the reverberated signal.**
Raising the Damp value cuts off the high frequencies of the reverb, thereby creating a smoother, warmer effect.

Send/Return switch

→ **Select whether you want the Amount knob to control the Send level or the Return level of the reverb effect.**
The picture below shows the different configurations schematically:



The schematic placement of the Amount knob in Send and Return mode, respectively.

← **To create gated or "reversed" reverb effects, set the switch to Return and modulate the Amount Modulation parameter from a modulation curve.**

Reverb On/Off button

→ **Click to switch on/off the Reverb effect section.**
If switched off, the signal is bypassed from the Reverb section.

Level



Level knob

- **Set the default level.**
 Range: - INF to + 12 dB.

! **If the Level is modulated by a modulation curve, the volume will increase above the default level, according to the modulation amount.**

In & Out switch

- **Select if the Level control should be routed before or after the rest of the effect sections in the signal chain.**

Master Controls



Dry/Wet

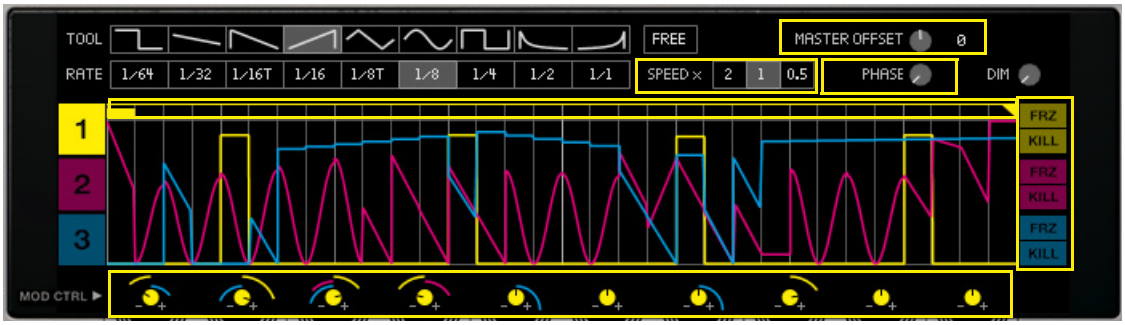
- **Set the balance between the dry input signal and wet signal for the entire effect chain.**
 At 0 the effects are completely bypassed and the input signal is passed through Synchronous unaffected.

Master Level

- **Set the master output level from Synchronous.**
 Range: - INF to + 12 dB.

About automation of display section parameters

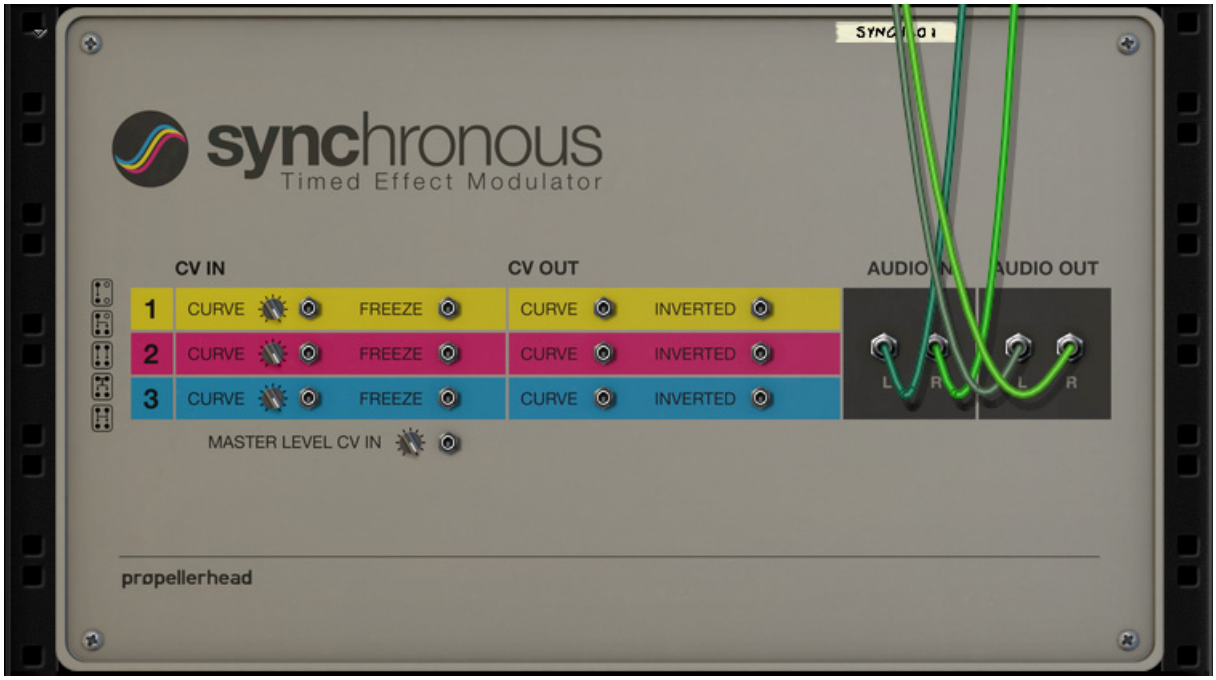
As with other Reason and Rack Extension devices it is possible to automate most of the panel parameters and control them from parameter automation lanes in the main sequencer. Besides the regular panel parameters in Synchronous, parameters in the display section can also be automated:



The parameters in the display section that can be automated are:

- **The Master Offset control.**
- **The Speed control for each modulation curve.**
- **The Phase control for each modulation curve.**
- **The Loop Locator for each modulation curve.**
- **The FRZ (Freeze) and Kill buttons for each modulation curve.**
- **The Modulation Control knobs for each modulation curve.**

Connections



CV In

Curve 1/2/3

These CV inputs accept bipolar control signals. Each input CV signal is added to the corresponding Modulation Curve signal and the resulting signal then modulates the assigned effect parameters.

The input signals can be attenuated with the corresponding attenuation knobs.

! **If the resulting modulation curve value should be negative (below zero), the assigned effect parameters are modulated with reversed polarity.**

Freeze 1/2/3

CV signals with levels >0 patched to these inputs will activate the Freeze function for the corresponding Modulation Curve. When the CV signals drops to 0 or below, the Freeze function is deactivated. See "FRZ (freeze) buttons" for more information.

Master Level CV In

A CV signal on this input can modulate the "Master Level" parameter. You can attenuate the input CV signal with the attenuation knob.

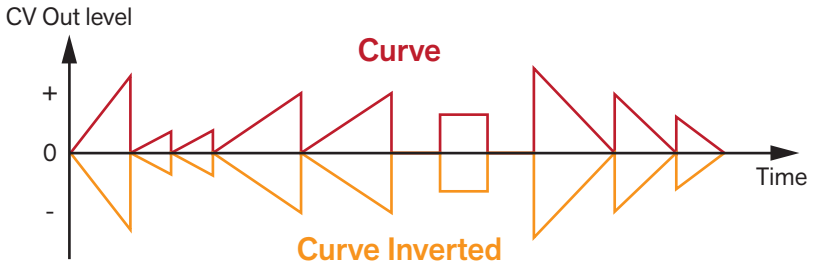
CV Out

Curve 1/2/3 and Curve 1/2/3 Inverted

These CV outputs send out positive unipolar control signal levels according to the shapes of Modulation Curves 1-3 respectively.

The “Inverted” CV outputs below each “regular” CV output send out inverted negative unipolar control signal levels according to the shapes of Modulation Curves 1-3 respectively.

The picture below shows a Curve signal (dark red) and its corresponding Curve Inverted signal (orange):



- **By patching one of the positive control signals to one destination and the corresponding inverted negative control signal to another destination, you are able to control the parameters in a “mirrored” fashion. For example, patch the positive CV signal to the Filter Cutoff modulation of a synth device and the negative signal to the Resonance modulation of the same synth.**

Audio In L&R

- **Patch the audio signals you want to process here.**
 If your input signal is in mono, connect only to the L (left) input.

Audio Out L&R

These are the stereo audio outputs.

Chapter 44
The MClass Effects



The MClass effects

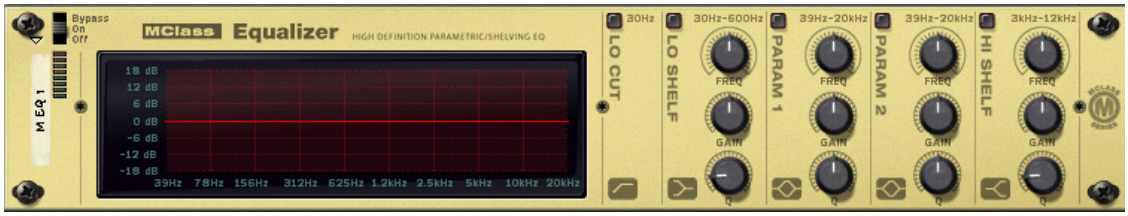
The MClass effects package consists of four effect devices, which are available as separate devices on the Effects palette in the Browser. There are also a number of Mastering Combi patches available in the Effects folder in the Factory Sound Bank. These mastering patches consist of all four MClass device types, patched together and organized as Combinator patches. The Mastering Combis are perfect to use as Master Insert FX to process the final mix. The individual MClass effects are as follows:

- **The MClass Equalizer - this has low and high shelving bands, two fully parametric bands and a low cut "anti-rumble" switch. See "The MClass Equalizer".**
- **The MClass Stereo Imager - this can be used to control the stereo width for the high and low frequency ranges separately. See "The MClass Stereo Imager".**
- **The MClass Compressor - this single band compressor features sidechain input and program-adaptive release. See "The MClass Compressor".**
- **The MClass Maximizer is a special limiter tailored for loudness maximizing without risk of clipping. See "The MClass Maximizer".**

Common effect device parameters

While the specific parameters for the MClass effect devices are described below, some features and procedures are common to all effect devices. Please, refer to "Common effect device features" for information about the Input meter, the Bypass/On/Off switch and Signal Flow Graphs on the effect devices.

The MClass Equalizer



The MClass Equalizer consists of two independent, fully parametric bands plus high and low shelving bands and a lo cut switch.

This is most often used as an insert effect, in mono or stereo.

Activating the separate EQ bands

The separate bands are organized as vertical strips on the panel, in the following order (from left to right): Lo Cut/Lo Shelf/Parametric 1/Parametric 2/Hi Shelf.

You activate the separate bands by clicking the button at the top of each strip (none of the bands are activated by default).

Lo Cut

The Lo Cut switch will simply cut frequencies below 30 Hz (by 12 dB/Octave). This is useful for removing low frequency "rumble".

- **When you are using the MClass Equalizer with a compressor or Maximizer, activating the Lo Cut switch prevents subsonic sound from "topping" the compressor/limiter, and allows them to operate as efficiently as possible.**

Parametric 1-2 parameters

A parametric equalizer will boost or cut frequencies around the selected frequency. The following parameters are available for both the parametric bands:

Parameter	Description
Frequency	This determines the center frequency of the EQ, i.e. at which frequency the level should be decreased or increased. The range is 39 Hz to 20 kHz.
Gain	Specifies how much the level of the selected frequency range should be boosted or cut. The gain range is ±18 dB.
Q	This governs the width of the affected area around the set center frequency. The higher the value, the narrower the affected frequency range.

Lo/Hi Shelf parameters

A shelving equalizer will boost or cut frequencies below or above the selected frequency.

The following parameters are available for the Lo/Hi Shelf bands:

Parameter	Description
Frequency	Frequencies below (Lo Shelf) or above (Hi Shelf) the selected frequency will be boosted or cut. • The Lo Shelf range is 30 Hz to 600 Hz. • The Hi Shelf range is 3 kHz to 12 kHz.
Gain	Specifies how much the level should be boosted or cut. The gain range is ±18 dB.
Q	This governs the slope of the shelving curve. The higher the value, the steeper the curve slope. High Q settings will also produce a "bump" in the opposite cut/boost direction at the set frequency.

About the graphic display

The graphic display to the left in the device panel shows the frequency response curve as set by the EQ parameters. This gives a visual feedback and helps you tailor the EQ settings.

The MClass Stereo Imager



The MClass Stereo Imager splits the signal into two frequency bands; "Hi" and "Lo" and allows you to widen or narrow the stereo image of each band. A typical application of the Stereo Imager is to widen the higher frequencies and narrow the lower frequencies. This will make the bass end "tight" whilst "opening up" the higher frequencies.

This is most often used as an insert effect in stereo.

! The MClass Stereo Imager does not create stereo from mono input! For the device to work properly it must connected with stereo in/out, and the input signal must contain a stereo audio signal.

Parameters

The following parameters are available:

Parameter	Description
X-Over Frequency	This determines the crossover frequency between the Hi and Lo band. Range is 100 Hz - 6 kHz. Frequencies below this will be affected by the Lo Width setting; frequencies above will be affected by the Hi Width setting.
Lo Width	This adjusts the stereo width for the Lo band. Turn anti-clockwise to narrow the stereo width (i.e. to make it more "mono"), and clockwise to widen the stereo image. Center position means no change from original signal. The "Active" LED indicates whether Low Width is activated or not. Note that for the Lo band, it is more common to narrow the stereo image, as the low frequency content in a mix is usually mixed center and can become less defined if widened.
Hi Width	This adjusts the stereo width for the Hi band. Turn anti-clockwise to narrow the stereo width (i.e. to make it more "mono"), and clockwise to widen the stereo image. Center position means no change from original signal. The "Active" LED indicates whether Hi Width is activated or not.
Solo switch	This allows you to listen to the Lo and Hi bands separately, for reference purposes. "Normal" is the standard operating mode.

Connections



Apart from standard L/R inputs and outputs, there are also "Separate" L/R outputs on the back panel. The Separate outputs can either carry the Lo or Hi band output, which is set by the switch beside the outputs. These outputs can be used to apply processing separately to either the Lo or Hi band.

- **If you set the Solo switch to "Lo" and the Separate output switch to "Hi", the device will operate as a basic crossover filter, delivering the Lo band signal from the main output and the Hi band signal from the Separate out.**

The MClass Compressor



This is a single-band compressor capable of everything from subtle compression to aggressive pumping effects. Like all dynamics processors it is best used as an insert effect.

The features include "soft-knee" compression for more musical and unobtrusive compression, program-adaptive release time and a sidechain input for de-essing and other dynamics processing. Additionally, you have a CV output, allowing you to have the amount of gain reduction control other Reason parameters.

Parameters:

Parameter	Description
Input Gain	The Input Gain controls the "drive" of the compression. This determines how much compression the signal will have in conjunction with the Threshold. Range: ±12 dB.
Threshold	This sets the level at which onset of compression occurs. When the input level is below the Threshold setting the signal is unaffected. When the input level exceeds the threshold, compression kicks in. In practice, this means that the lower the Threshold setting (and the higher the Input Gain), the more compression will be applied. Range: -36 dB to 0 dB
Soft Knee	Normally signals above the threshold will be compressed immediately at whatever ratio is set. This can be very noticeable, especially when using high compression ratios. When Soft Knee is activated, the onset of compression will be more gradual, producing a less drastic result.
Ratio	This lets you specify the amount of gain reduction applied to the signals above the set threshold. The Ratio can be set from 1:1 (no reduction) to ∞:1 (Infinite).
Gain meter	This shows the amount of gain reduction (in dB).
Solo Sidechain	This allows you to monitor the signal connected to the sidechain input (see below).
Attack	This governs how quickly the compressor will apply its effect when signals rise above the set threshold. If you raise this value, the response will be slower, allowing more of the signal to pass through the compressor unaffected. Typically, this is used for preserving the attacks of the sounds. Range: 1ms to 100ms.
Release	When the signal level drops below the set threshold, this determines how long it takes before the compressor lets the sound through unaffected. Set this to short values for intense, "pumping" compressor effects, or to longer values for a smoother change of the dynamics. Range: 50ms to 600ms.
Adapt Release	When this is used, set Release to the time you want for short peaks - when longer peaks occur, the Release time is automatically increased.
Output Gain	This controls the output gain and can be used to compensate for the gain reduction caused by compression. Range: ±12 dB.

About the Sidechain inputs

- **When a signal is connected to the Sidechain inputs, it is this signal that will trigger the compression.**
 This signal is not passed to the compressor output, and will thus not be heard (unless Sidechain Solo is activated).
- **The signal connected to the normal L/R inputs is the signal that will be processed by the compressor (as usual).**

Here follows two examples of how you can use sidechain processing:

Example 1 - using the Sidechain inputs to create ducking effects

"Ducking" is when the level of one signal is reduced by the presence of another signal. A typical application is to automatically lower the level of a musical bed when a voice-over starts, and to automatically bring the level up when the voice-over stops. To set this up, we can name the musical bed in the example "Device A", and the voice-over "Device B". Proceed as follows:

1. **Connect Device A and Device B to separate channels on a mixer device.**
 In this example the signal produced by Device A should be continuous, and the signal produced by Device B should be intermittent, i.e. it should contain both silent passages and signal passages.
2. **Select Device A and then select an MClass compressor from the Create menu.**
 The effect will be auto-routed as an insert effect to Device A.
3. **Connect a Send on the mixer device to the Sidechain inputs on the MClass compressor.**
 As soon as anything is connected to the Sidechain inputs, the "Active" indicator lights up on the device panel. The compressor will no longer react to the signal produced by Device A.



4. **Turn up the corresponding AUX Send level for Device B on the mixer.**
 This means that the Device B signal now feeds both the mixer's input, and the sidechain input on the compressor, which in turn triggers the gain reduction.
5. **If you now start playback of both devices, the level of Device A will be lowered whenever Device B sounds, and be raised again when Device B stops.**
 The amount of gain reduction, how quickly it lowers the level, and the time it take for the level to return to normal again is determined by the corresponding Gain/Threshold/Ratio and Attack/Release parameters.

Example 2 - using the Sidechain inputs to create frequency sensitive compression

By inserting an equalized signal to the sidechain inputs you can make the compression more or less sensitive to a certain frequency range. A typical application of this is "de-essing" - where harsh "S"-sounds in vocal material is reduced or eliminated.

Frequency sensitive compression is set up as follows:

1. **Hold down [Shift] and create an instrument device.**
 Pressing [Shift] means no auto-routing connections to/from the device are made.
2. **Hold down [Shift] and create a MClass Equalizer.**
3. **Hold down [Shift] and create a MClass Compressor.**
4. **Create a Spider Audio Merger and Splitter device.**
5. **Connect the outputs of the instrument device to the A and B inputs on the Spider.**

6. **Route one pair of the split outputs of the Spider to the MClass Equalizer inputs.**
7. **Route the Equalizer outputs to the Sidechain inputs on the MClass Compressor.**
8. **Route another pair of the split outputs of the Spider to the MClass Compressor.**
 Now, the compressors normal audio inputs are fed the unprocessed signal, and the sidechain inputs are fed the equalized signal.



9. **Next, route the outputs of the Compressor to inputs on a mixer device.**
10. **Activate the Solo Sidechain button on the compressor's front panel.**
 You will now only hear the equalized signal.
11. **Now use the parameters on the MClass Equalizer to boost the frequencies you wish should trigger the gain reduction and cut the frequencies you wish to avoid triggering the gain reduction.**
 You can use rather extreme eq settings - the signal will not be heard when Solo Sidechain is deactivated anyway. E.g. for de-essing you should separate and boost the offending "S" frequencies as much as possible.
12. **Deactivate Solo Sidechain when you have finished tweaking the equalizer.**
 Now, the compressor will be more sensitive to the frequency area you tuned in with the equalizer, and thus react more to these frequencies. Note, however, that the whole signal will still be compressed - not just the boosted frequencies - so in case of de-essing you should usually use fast Attack and Release settings so that the gain reduction does not affect the rest of the program too much.

CV Outs

On the back of the MClass Compressor you can find a "Gain Reduction" CV out connector. This can be used to modulate other parameters with the amount of gain reduction applied by the compressor. This means that the compressor works as an envelope follower. You could for example have the audio signal level control pan in a mixer or a synth parameter.

The MClass Maximizer



This is a loudness maximizer, a special type of limiter which can significantly raise the perceived loudness of a mix without risk of hard clipping distortion. Features include a 4 ms look ahead function for “brick wall” limiting and a Soft Clip function.

The MClass Maximizer should be used as an insert effect, and is designed to be placed at the end of the signal chain between the mixed final output and the I/O device.

Parameters

Parameter	Description
Input Gain	The Input Gain sets the basic volume of a mix. If this is set very high, you should use Look Ahead mode or the Soft Clip function to eliminate the risk of hard clipping distortion. Range: ±12 dB.
Limiter On/Off	This turns the Limiter section on or off.
Look Ahead On/Off	If activated, this will introduce a very short delay (4 ms) to the signal. This delay is used to detect peaks in the signal before they actually occur. If high peaks are detected the limiter is “ready for them” and gain reduction is applied to transparently control the peaks.
Attack (Fast/Mid/Slow)	This governs how quickly the Limiter will apply its effect. If set to Fast with Look Ahead activated (and the Output Gain is set to 0 dB) you will get “brick wall” limiting - no signal peaks over 0 dB will pass.
Release (Fast/Slow/Auto)	This determines how long it takes before the Limiter lets the sound through unaffected. If Auto is activated, the Release time will automatically adapt to the program material.
Output Gain	This controls the output gain and should normally be set to 0 dB.
Soft Clip On/Off	If this is activated, it also acts a 0 dB brick wall limiter but in a slightly different way. The signal will be “soft-clipped” which adds a pleasant, warm sounding distortion to the signal. It can be used simply to get this effect, or as a safeguard against hard clipping distortion if Look Ahead with Mid or Slow attack settings are used (or if Look Ahead is deactivated).
Soft Clip Amount	This controls the amount of soft-clipping distortion. Note that if Soft Clip is on but the Amount is set to zero, the distortion will be like hard clipping, and thus less pleasing to the ear.
Output level meter (Peak/VU)	This is a more detailed meter than found on the mixer. You can switch the meter characteristics between Peak (faster response to peaks) and VU mode (average levels).

Chapter 45
Half-Rack Effects

Common effect device features

While the specific parameters for each effect device are described below, some features and procedures are common to all effect devices:

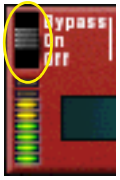
The Input meter



This shows the level of the incoming audio signal, giving you an indication of which devices are active, connected and playing. However, you don't need to worry about clipping in effect devices, even if the meter goes into the red.

The Power/Bypass switch

This is located in the upper left corner of each effect device. The switch has three modes, according to the following figure:



Mode	Description
Bypass	In this mode, the input signal is passed directly to the audio output, without being affected by the effect device. This is useful when the effect device is connected as an insert effect, and you want to compare the effect sound with the dry sound.
On	This is the default mode, in which the device processes the incoming signal.
Off	In this mode, the effect device is turned off and neither dry nor effect sound is sent out. This is useful when the device is connected as a send effect and you want to turn it off temporarily.

About making settings

You adjust effect parameters using the regular editing techniques.

- **A quick way to reset the parameters to their default values is to hold down [Ctrl](Win) or [Cmd](Mac) and click the corresponding knob.**

About Connections

- **All effect devices have stereo inputs and outputs, and can be connected as send effects or as insert effects.** However, some effects are best used as one of these only. This is stated for each effect on the following pages. See also the section about the signal flow graphs below.
- **Most of the effect devices have one or several CV inputs on the back panel.** These allow you to control various effect parameters in real-time, from another device in the rack. See "CV/Gate signals" for details about routing CV.

The Signal Flow graphs



On the back of each effect device, you will find two or three small “graphs”. These indicate how the effect device handles mono and stereo signals, depending on the connections. The selection of graphs for a device tells you how it should be used, according to the following rules:

Graph	Description
	Can be connected as a mono-in, mono-out device. (Of course, all effect devices *can* be connected in mono. However, if this graph isn't shown for a device, this means that a mono-in, mono-out connection may not give the proper results).
	Can be connected as a mono-in, stereo-out device. This means that the device creates some sort of stereo effect (e.g. a reverb) or a mono effect that can be panned.
	If you connect both inputs and outputs in stereo, the two sides will be processed independently (dual mono processing).
	If you connect both inputs and outputs in stereo, the two sides are summed before the effect processing. However, the actual effect is in stereo (and the dry signal will remain in stereo, if it is passed through the effect).
	“True stereo” processing, or “stereo in - stereo out” processing. When you connect the inputs in stereo, each channel in the effect uses the signal information from both inputs. However, the inputs are not summed - the two channels are processed differently. This mode is available on the RV7000 Advanced Reverb - see “RV7000 Mk II Advanced Reverb”.

DDL-1 Digital Delay Line



This is a mono delay (where the output can be panned in stereo) that can be synchronized to the song tempo. The delay can be used as a send effect or an insert effect.

Parameters

Parameter	Description
Delay time	The display to the left on the device panel shows the delay time, either as note value steps (based on the sequencer tempo and the Step Length parameter) or in milliseconds, depending on the setting of the Unit switch. The maximum delay time is two seconds (2000 ms) while the maximum number of steps is 16. Note that if the tempo is low, you may reach the maximum delay time at a lower number of steps than 16 (in which case raising the steps value will not make any difference).
Unit	This is where you select whether you want a tempo-based delay ("Steps" mode) or a free time delay ("MS" mode). In the Steps mode, you specify the delay time in note value-based steps. This means that if you change the tempo in the transport panel, the delay will maintain its rhythmic relation to the music (provided that the resulting delay time doesn't reach the maximum value). This mode is useful for creating rhythmic patterns. If you change the tempo when using the delay in MS mode, the delay time will remain the same. See also the note about switching Unit modes below.
Step length	Governs whether each step in Steps mode should be a sixteenth note (1/16) or an eighth triplet note (1/8T).
Feedback	Determines the number of delay repeats.
Pan	Pans the delay effect to the left or to the right.
Wet/Dry	If you are using the delay as an insert effect, you use this parameter to adjust the balance between the unprocessed audio signal (dry) and the delay effect (wet). If the delay is used as a send effect, this should be set all the way to wet only, since you can control the balance by using the AUX send controls in the Mixer.

CV Inputs

The following CV inputs are available on the back panel of the device:

- **Pan CV.**
 This allows you to control the panning of the delay signal. Connect an LFO to this for moving delay effects, or use a Matrix pattern to simulate random delay panning.
- **Feedback CV.**
 This allows you to control the amount of feedback (the number of delay repeats) from another device. Useful for dub-type echoes on certain beats or notes only.

Switching between Unit modes

When you switch between the two Unit modes (Steps and MS), the following rules apply:

- **If you switch from Steps mode to MS mode, the delay will be set to the same actual delay time as was used in the Steps mode.**
 This means that you can set up an exact rhythmic delay in Steps mode, and then switch to MS mode to adjust it slightly.
- **If you switch from MS mode to Steps mode, the delay is reset to the previously used Steps value.**

CF-101 Chorus/Flanger



The CF-101 is a combined chorus and flanger effect. It adds depth and movement to the sound by adding a short modulated delay to the audio signal. The delayed signal is then mixed with the original (either in the effect device or manually by you - see below). The CF-101 can be used as an insert or send effect.

Parameters

Parameter	Description
Delay	This is a manual control for the delay time used to create the chorus/flanger effect. Usually, flanger-type effects use fairly short delay times while chorus-type effects use medium long delays.
Feedback	This governs the amount of effect signal fed back to the input, which in turn affects the intensity and character of the effect. Turning this to the extreme left (negative feedback) or right (positive feedback) produces different flanger effects with a pronounced resonance "tone", while keeping it in between produces a more gentle chorus effect.
LFO Rate	This is the frequency of the LFO modulating the delay time. The higher the value, the faster the sound will oscillate.
LFO Sync	This button lets you activate/deactivate LFO sync. When it is activated, the frequency of the LFO is synchronized to the song tempo, in one of 16 possible time divisions. The LFO Rate knob is then used for setting the desired time division. Turn the knob and check the tooltip for an indication of the time division.
LFO Mod Amount	This determines the depth of the LFO modulation, i.e. by how much the delay time should be modulated. If you set this to 0, the effect will be "frozen" (most effective if you add some feedback).
Send Mode	This determines whether the delayed signal and the dry signal should be mixed in the effect device or not. If you use CF-101 as an insert effect, you should turn this off - the device will then output a mix of the dry signal and the modulated delay signal. If you use the device as a send effect, you should activate Send mode. Then, the device will only output the modulated delay signal, allowing you to mix it with the dry signal using the AUX send controls in the mixer. See also the note below about using the CF-101 as a vibrato effect!

CV Inputs

The following CV inputs are available on the back panel of the device:

- **Delay CV.**
 Allows you to control the delay time from another device. This may give best results if you turn off the LFO modulation in the device (turn LFO Mod Amount to zero). For example, by controlling the delay parameter from a Matrix, you can create "stepped flanger" effects, in sync with the tempo.
 - **If you use the Delay CV input for "playing" the feedback tone, note that a higher delay value gives a lower pitch.**
- **Rate CV.**
 Lets you control the rate of the modulating LFO from another device.

About Stereo and Mono connections

The CF-101 can be connected using mono or stereo inputs, as shown by the graphs on the back panel. Note the following:

- **A "moving" stereo effect is only obtained when you use a mono input and stereo outputs.**
 With a stereo input, the two sides are processed in parallel, maintaining the stereo image of the original sound.

- **When you are using a mono input and stereo outputs, there will only be a stereo effect if the internal LFO is used.**
 If you turn LFO Mod Amount to zero, both stereo outputs will carry the same signal (mono). This is because the "fake stereo" effect is produced by inverting the modulation for one of the output channels.

Tip: Using the CF-101 as a vibrato effect

The Send mode is intended for when using the CF-101 as a send effect. In this mode, the device will only output the modulated delay signal - you get the actual "chorusing" by mixing this signal with the dry, unprocessed signal in the Mixer.

However, if you activate Send mode while using the device as an insert effect, the result will be a pitch modulated version of the original sound - in short, a vibrato effect. Along with a little feedback, this can be used for special effects.

Spider Audio Merger & Splitter

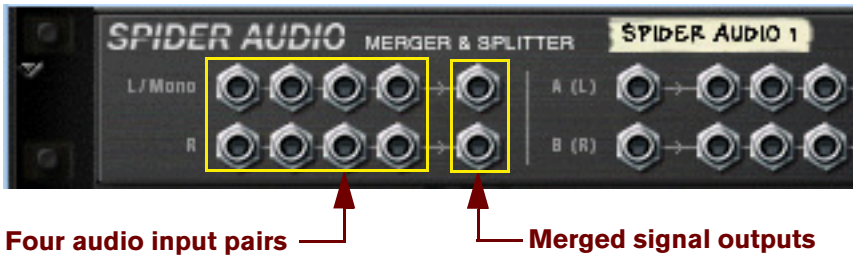


The Spider Audio Merger & Splitter is not an effect device, but a utility. It has two basic functions:

- **To merge up to four audio input signals into one output.**
- **To split one audio input signal into four outputs.**

There are no controls on the front panel of this device, only signal indicators.

Merging audio



On the back panel of the Spider are several audio connectors. The left half of the panel contains four stereo audio input connectors, and to the right of these, one merged stereo output.

- **The principle is simple; all audio signals connected to any of the four inputs will be merged and output via the output connectors.**
 If you connect a mono signal (to a L/Mono input, with nothing connected to the corresponding R input) it will be output on both merged outputs. This way you can merge stereo and mono signals freely.
 If you connect a signal to the R input only (with nothing connected to the corresponding L/Mono input) it will be output on the R output only.

Practical uses of merging audio

There are many practical uses of merging audio signals together, for example:

- **Process several audio signals with the same insert effect(s).**
 Perhaps you want to process certain channels in a mix with the same compressor, or use one ECF-42 to filter a group of instruments in a mix. You can also set up a chain of insert effects and process the merged signals.
- **Sub-grouping signals.**
 It may be practical to control several audio signals using one channel strip in the Mixer.
- **Use merged signals as either carrier or modulator source for the BV512 Vocoder.**
 You could use several sounds as carrier signal, or modulate the carrier with several signal sources.

Splitting audio

The right half of the back panel contains two signal splitters, labeled “A (L)” and “B (R)”. The two splitters work independently, in the following way:

- **The signal fed to the input connector will be simultaneously output by all four outputs.**
 For splitting stereo signals you simply use both splitters with “A” for the left channel and “B” for the right channel.



Practical uses of splitting audio

There are many practical uses of splitting audio signals - here a few examples:

- **Create “pseudo” stereo effects from mono signals.**
 For example, you could route the mono output of a Subtractor to the Spider and then send two split outputs (from the same row) to different effects and on to different Mixer channels panned left and right.
- **It provides a way to instantly switch between (or mix) different variations of the same signal.**
 This is a neat way of applying “spot effects” in a mix. An instrument output is split and sent to three different combinations of insert effect processing. The outputs from the three effects are routed to separate channels in the Mixer, which could in turn have different send effects, eq, etc. You then have three different variations of the same signal that can be easily switched in and out of the mix for drastic sonic changes - or combined for huge layered sounds.

Spider CV Merger & Splitter

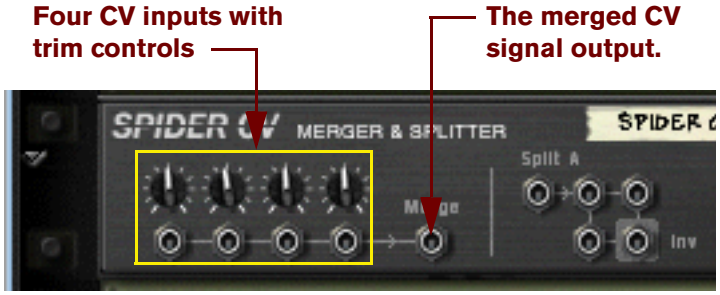


The Spider CV Merger & Splitter is not an effect device, but a utility. It has two basic functions:

- **To provide one merged CV output from up to four CV input sources.**
- **To split CV or Gate inputs into several outputs.**
 Two inputs, A and B, are provided, each with four outputs, where one of the outputs will invert the polarity of the control signal. One reason for having two splitable inputs is to make it possible to split Gate and Note CV, to control several instrument devices with one Matrix for example.

There are no controls on the front panel of this device, only CV signal indicators. The four horizontal indicators light up to indicate signals connected to the corresponding merge input. The two indicators to the right indicate signals connected to the corresponding split inputs.

Merging CV



On the back panel of the Spider there are several CV connectors. The left half of the panel contains four CV/Gate input connectors with associated trimpots, and to the right of these, one merged CV output.

- **The merged CV output will produce a CV signal that represents the "sum" of all connected CV inputs.**

A few things to note:

- **Gate CV signals typically trigger notes or envelope cycles and are normally routed to a Gate input.**
- **CV signals typically control note pitch or for modulating parameters and are typically routed to CV Note or Modulation inputs.**

There are no strict rules involved, but the facts mentioned above means that it is generally better to stick to using either Gate CV signals or CV signals but not a mixture when merging. simply because the CV/Gate signals usually go to different input destinations.

For instance, merging Note CV and Gate CV from a Matrix does not make much sense if you want to use Matrix to play melodic patterns via the Sequencer Control inputs of an instrument device. There would only be one merged output whereas the instrument device would need a separate Gate and Note CV signal to work properly.

Practical uses of merging CV

The practical applications of merging CV are perhaps less obvious compared to splitting CV. But there are numerous applications for a merged CV control output, a few of which are listed below:

- **You can create interesting modulation effects by merging several Modulation outputs from LFO's and other CV modulation sources.**
 For example, merging the Modulation outputs from several LFO's would produce a "mixed modulation" output. This merged output signal could be likened to a "super LFO" capable of generating several modulation cycles simultaneously, each with a different waveform and modulation rate! In addition to this, by using the trim control for each CV input, you have full control over the amount of modulation applied by each LFO.

 The above example could of course also include Curve CV outputs from a Matrix or Mod Outs from Malström etc., in short any CV Modulation output.

- **Use the ECF-42 Filter to apply envelope controlled filter effects.**
 This can create the sound of "synthesized" percussion, and other interesting effects.

This is done using the following method:

1. **Connect the audio outputs of a Redrum to a ECF-42 Envelope controlled filter.**
2. **Connect the Gate outputs from up to 4 Redrum drum channels to the merge inputs of a Spider CV.**
3. **Route the merged output to the Env Gate input on the ECF-42.**
 If you add a touch of velocity the connected Gate signals will trigger the ECF-42 filter envelope. Again, the trim pots on the Spider allows you to adjust the amount of filter envelope applied.

- **Create an "arpeggiator" using two Matrix devices and the Spider CV Merger & Splitter.**

By merging the Note CV output from one Matrix with a Curve CV output of another Matrix, you can transpose the Matrix pattern in real-time, a bit like an arpeggiator.

1. **Create a Subtractor and a Matrix device.**
 Connect the Matrix Note and Gate CV outputs to the Subtractor Sequencer CV and Gate inputs, respectively.
2. **Program a pattern for the Matrix.**
 In the following text this is referred to as "Matrix 1".

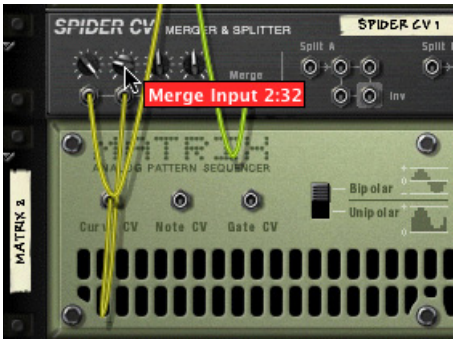
3. **Now create a Spider CV and a second Matrix device and connect them as in the picture below.**



Note that the Note CV output from Matrix 1, and the Curve CV output of Matrix 2 should be connected to the Spider. The merged output is connected to the Sequencer Control Note CV input on the Subtractor.

4. **On the Spider CV, turn the trimpot for the input connected to the Note CV output fully to the right.**
 This setting will retain the correct pitch relationship for the notes played by the pattern.

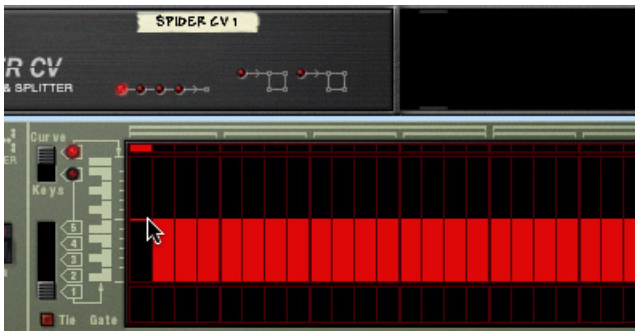
5. **On the Spider CV, turn the trimpot for the input connected to the Curve CV output to "32".**
 This will produce a Curve CV output that corresponds to semitone steps.



6. **Set the Curve type switch to “Bipolar” on the back of the second Matrix (Matrix 2).**
7. **Flip the rack around so that the front panels are showing, and make the following settings for the “Matrix 2”:**

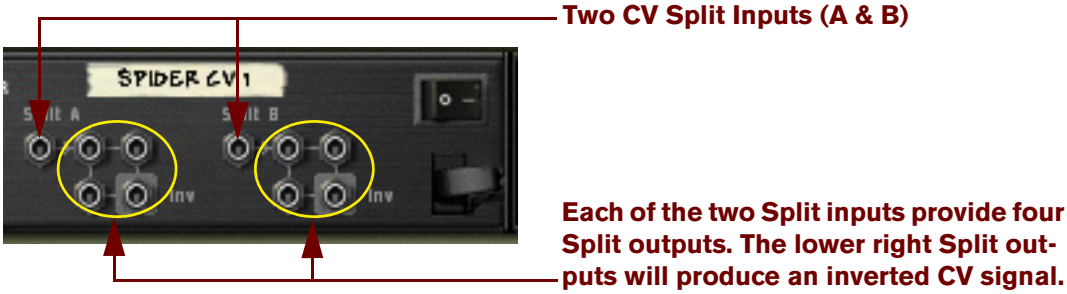
- **Set the number of steps to “1”.**
- **Set the Curve/Keys switch to “Curve”.**

8. **Adjust the Matrix 2 curve for step 1 (the only step used) so that it is in the middle of the bipolar curve as the picture shows.**



9. **If you now activate Play from the transport, the pattern you programmed for Matrix 1 is played back. By carefully adjusting the Matrix 2 Curve step 1 up or down the Matrix 1 pattern is transposed in semitone steps.**
 By programming different values for the “pattern” played by Matrix 2 and saving them in different pattern locations, you can use the Pattern selectors to transpose the Matrix 1 pattern to different keys!

Splitting CV



On the right half of the back panel you will find two split inputs “A” and “B”, each with four output connectors. The signal connected to a Split input will be output by all four corresponding outputs, where one is inverted.

Practical uses of splitting CV

There are many practical uses of splitting CV signals - here are a few examples:

- **Connecting the CV Note and CV Gate outputs from a Matrix to Split Input A and B, allows you to connect the Matrix to several instrument devices.**
 Simply route the CV and Gate outputs to the corresponding Sequencer Control CV/Gate inputs on the instrument devices. Although this could also be done by copying the Matrix Pattern data to several sequencer tracks and routing the outputs to the desired devices, the advantage by using Split is if you are editing Matrix pattern data this will be immediately be reflected in all the connected devices, without any copy/paste operations.
- **Splitting modulation outputs from LFO’s, Curve CV data etc. allows you to apply modulation from one source to several parameters.**
 By using the inverted output, you can create interesting modulation crossfades, where one parameter value rises and another parameter value is lowered for example.

RV-7 Digital Reverb



! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Reverb adds ambience and creates a space effect. Normally, reverb simulates some kind of acoustic environment such as a room or a hall, but you could also use it as a special effect.

- **The Reverb device can be used as a send effect or an insert effect.**
 If several devices uses the same type of reverb, you should connect the reverb as a send effect, to conserve computer power.

Parameters

The display to the left on the panel shows the selected reverb algorithm - the general type of reverb. By clicking the arrow buttons you can change algorithm, with the following options available:

Algorithm	Description
Hall	Emulates a fairly large, smooth hall.
Large Hall	Emulates a larger hall, with pronounced pre-delay.
Hall 2	A hall reverb with a brighter attack than “Hall”.
Large Room	Emulates a large room with hard early reflections.
Medium Room	Emulates a medium-sized room with semi-hard walls.
Small Room	A smaller room, suitable for “drum booth”-type reverbs.
Gated	A gated reverb, that is abruptly cut off.
Low Density	A thinly spaced reverb, where you clearly can here the individual echoes. Useful for strings and pads and as a special effect.
Stereo Echoes	An echo effect with the repeats alternating between stereo sides.
Pan Room	This is slightly similar to “Stereo Echoes”, but the echo repeats have soft attacks.

► **If you need to conserve computer power, try using the Low Density algorithm. This uses much less power than the other algorithms.**

The selected reverb algorithm can be tweaked using the parameters on the device panel:

Parameter	Description
Size	Adjusts the emulated room size. Middle position (value 0) is the default size for the selected algorithm. Lowering this parameter results in a closer and gradually more “canned” sound. Raising this parameter results in a more spacey sound, with longer pre-delay. For the “Stereo Echoes” and “Pan Room” algorithms, the Size parameter adjusts the delay time.
Decay	This governs the length of the reverb effect. Middle position is the default decay time for the selected algorithm. Note: Decay is not used for the “Gated” algorithm.
Damp	Raising the Damp value cuts off the high frequencies of the reverb, thereby creating a smoother, warmer effect.
Dry/Wet	If you are using the reverb as an insert effect, you use this parameter to adjust the balance between the unprocessed audio signal (dry) and the effect (wet). If the reverb is used as a send effect, this should be set all the way to wet only, since you can control the balance by using the AUX send controls in the Mixer.

CV Inputs

You can control the Decay parameter via the CV input on the back of the Reverb device.



D-11 Foldback Distortion



! Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.

The D-11 is a simple but effective distortion effect, capable of producing anything from just a whisper soft touch of distortion, to complete thrashing. This effect is most often used as an insert effect.

Parameters

The distortion has the following parameters:

Parameter	Description
Amount	This controls the amount of distortion. The higher the value, the more distortion.
Foldback	This adjusts the character of the distortion by introducing foldback, which makes the waveform more complex. The default value is in the middle position. This produces a "flat" clipping distortion, which is the most common type. Lowering the parameter makes the sound rounder and more gentle, raising it makes the sound sharper and more evil.

CV Inputs

On the D-11 you will find a CV input for controlling the Amount parameter. This can produce very drastic effects, especially if you control parameters in the instrument device (such as filter frequency and resonance) at the same time.

ECF-42 Envelope Controlled Filter



! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

The ECF-42 is a multimode filter with a built in envelope generator. It is mainly designed to be used together with pattern devices to create pattern controlled filter and envelope effects, but it can also be triggered via MIDI, or used as a "static" filter for shaping the sound of an instrument device or a whole mix.

Usage

The Envelope Controlled Filter is best connected as an insert effect. However, unlike the other effects it is not a pure "stand-alone" device. To make the most of the ECF-42, you need either CV/Gate from an external device or MIDI notes from a sequencer track.

- **If you connect a device to the ECF-42 using audio inputs/outputs only, it will simply act as a filter with no velocity or envelope modulation.**
 Hence, all filter parameters are "static", unless you manually turn the knobs or automate them in the sequencer.
- **Connecting a gate signal to the Env Gate input on the back panel of the device allows you to trigger the envelope generator for the filter.**
 Note that the ECF-42 envelope generator is not triggered by the audio itself - the envelope parameters won't do anything unless the device receives gate signals.
- **By putting the ECF-42 in a Combinator (so it can receive MIDI), you can have the envelope triggered by MIDI notes on the track.**
 The envelope is affected by the position, length and velocity of the MIDI notes (but not by their pitch).

► **If you are unfamiliar with basic filter and envelope parameters, please refer to "Envelopes - General" in the Subtractor chapter for a description of these.**

The Filter Parameters



The ECF-42 filter section has the following parameters:

Parameter	Description
Mode	This button sets the desired filter mode. Three modes are available: 24dB/octave lowpass, 12dB/octave lowpass and 12dB/octave bandpass.
Freq	This is the filter cutoff frequency. When using the ECF-42 in "static" mode (without triggering the envelope), this parameter adjusts the frequency content of the sound. When using the envelope, the Freq parameter serves as the start and end frequency for the filter sweep.
Res	This is the filter resonance. Raising this produces a more extreme, "synthy" effect.
Env Amt	Determines how much the filter frequency should be affected when the envelope is triggered. The higher the value, the more drastic the effect. Note though, that if the Freq parameter is set high, raising the Envelope Amount will not make any difference over a certain value! This is because the filter is already fully opened - try lowering the Freq parameter in that case.
Velocity	This parameter determines how much the gate velocity value should affect the envelope amount.

The Envelope Parameters



This is a standard envelope generator with Attack, Decay, Sustain and Release parameters. It is triggered by a gate signal connected to the Env Gate input on the back panel, or by MIDI notes on a sequencer track connected to the ECF-42. The parameters have the following functionality:

Parameter	Description
A (Attack)	When the envelope is triggered, this is the time it takes before the envelope signal reaches its max value.
D (Decay)	After reaching its max value, this is the time it takes for the envelope signal to reach the sustain level.
S (Sustain)	If the gate remains open (or the MIDI note is held), the envelope signal will remain on this level.
R (Release)	When the gate is closed (gate CV goes back to 0) or the MIDI note ends, this is the time it takes for the envelope signal to drop from its current value to the start value (set by the Freq parameter).

- **The Gate indicator lights up when the device receives a signal to the Env. Gate input on the back panel or a MIDI note from a sequencer track.**

CV/Gate Inputs

On the back panel of the ECF-42, you can find the following CV/Gate inputs:

- **Freq CV.**
 Use this for controlling the filter frequency from another device. For smooth filter modulation, try connecting an LFO to this input.
- **Decay CV.**
 For controlling the envelope decay parameter from another device.
- **Res CV.**
 Allows you to control the filter resonance from another device. Can be very effective in combination with filter frequency sweeps.
- **Env. Gate.**
 This is where you connect a gate signal (e.g. from a Matrix or Redrum device) for triggering the envelope.

PH-90 Phaser



The PH-90 Phaser is a classic phaser effect with some special features for fine-tuning the sound. It can create the classic sweeping phaser sounds suitable for pads or guitars, but also more extreme effects if you like. The phaser is best used as an insert effect.

Theory

A phaser works by shifting portions of the audio signal out of phase, and then adding the processed signal back to the original one. This way, narrow bands of the frequency range ("notches") are filtered out. When these frequencies are adjusted, a sweeping phaser sound is created.

The PH-90 is a four-stage phaser, which means that there are four "notches" in the frequency response curve (this is a little like using four notch filters with different filter frequencies - see "Notch" in the Subtractor chapter for an explanation of notch filters).

When the phaser frequency is adjusted (manually or by the built-in LFO), these notches will move in parallel in the frequency spectrum. Furthermore, you can adjust the distance between the notches (Split) and their Width. Adding feedback raises the filter gain just below each notch in the frequency range, creating a more pronounced effect.

Parameters

Parameter	Description
Frequency	Sets the frequency of the first notch. Adjusting this will move the other notches correspondingly. This is the parameter modulated by the LFO to create phaser sweeps.
Split	This adjusts the distance between the notches in the frequency range, thereby changing the character of the effect.
Width	Determines the width of the notches. Raising the Width deepens the effect and simultaneously makes the sound more hollow and thin. This will also have an effect on character of the feedback "tone".
LFO Rate	This is the speed of the LFO modulating the frequency parameter. The higher the value, the faster the phaser sweeps.
LFO Sync	This button lets you activate/deactivate LFO sync. When it is activated, the frequency of the LFO is synchronized to the song tempo, in one of 16 possible time divisions. The LFO Rate knob is then used for setting the desired time division. Turn the knob and observe the tooltip that appears for an indication of the time division.
LFO Freq. Mod	This determines the depth of the LFO modulation, i.e. by how much the frequency parameter should be modulated. If you turn this to zero, the effect will be a static, formant-like sound (most effective if you add a little feedback).
Feedback	This is similar to the resonance control on a filter. Raising the feedback gives a more pronounced "tone" in the effect. For "singing" phaser sounds, try raising this to the maximum.

CV Inputs

The following CV inputs are available on the back panel of the device:

- **Freq CV.**
 Adjusts the frequency parameter. Use this e.g. for creating envelope controlled phasing (preferably with LFO Freq. Mod turned off in the device).
- **Rate CV.**
 Lets you control the speed of the modulating LFO from another device.

About Stereo and Mono connections

The PH-90 can be connected using mono or stereo inputs, as shown by the graphs on the back panel. Note the following:

- **A "moving" stereo effect is only obtained when you use a mono input and stereo outputs.**
 With a stereo input, the two sides are processed in parallel, maintaining the stereo image of the original sound.
- **When you are using a mono input and stereo outputs, there will only be a stereo effect if the internal LFO is used.**
 If you turn LFO Mod Amount to zero, both stereo outputs will carry the same signal (mono). This is because the "fake stereo" effect is produced by inverting the modulation for one of the output channels.

UN-16 Unison



! Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.

The UN-16 simulates the sound of several detuned voices playing the same notes simultaneously. The voices are individually slightly delayed and also pitch modulated by low frequency noise. This produces a rich chorus effect with the voices spread across the stereo field (given that stereo outputs are used).

The UN-16 can be used as an insert effect or a send effect.

Parameters

Parameter	Description
Voice Count	This switch sets the number of voices for the effect; 4. 8 or 16.
Detune	This sets the amount of detuning for the voices. Turn clockwise for stronger detuning effects.
Dry/Wet	If you are using the UN-16 as an insert effect, you use this parameter to adjust the balance between the unprocessed audio signal (dry) and the effect (wet). If the UN-16 is used as a send effect, this should be set all the way to wet only, since you can control the balance by using the AUX send controls in the Mixer.

CV Input

One CV input is available on the back panel of the device. This controls the Detune parameter.

COMP-01 Auto Make-up Gain Compressor



! Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.

The COMP-01 compressor levels out the audio, by making loud sounds softer. To compensate for the volume loss, the device has an automatic make-up gain, that raises the overall level by a suitable amount. The result is that the audio levels become more even and individual sounds can get more "power" and longer sustain.

The COMP-01 should be used as an insert effect, either for a single instrument device or for a whole mix (e.g. inserted between a Mixer device and the Hardware Interface). There are no CV inputs for this device.

Parameters

Parameter	Description
Ratio	This lets you specify the amount of gain reduction applied to the signals above the set threshold. The value is expressed as a ratio, from 1:1 (no reduction) to 16:1 (levels above the threshold are reduced by a factor 16).
Threshold	This is the threshold level above which the compression sets in. Signals with levels above the threshold will be affected, signals below it will not. In practice, this means that the lower the Threshold setting, the more the compressor effect.
Attack	This governs how quickly the compressor will apply its effect when signals rise above the set threshold. If you raise this value, the response will be slower, allowing more of the signal to pass through the compressor unaffected. Typically, this is used for preserving the attacks of the sounds.
Release	When the signal level drops below the set threshold, this determines how long it takes before the compressor lets the sound through unaffected. Set this to short values for intense, "pumping" compressor effects, or to longer values for a smoother change of the dynamics.
Gain meter	This shows the amount of gain reduction or increase (in dB), caused by the combined compression and make-up gain.

PEQ-2 Two Band Parametric EQ



! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

While there is a simple two-band shelving equalizer available for each channel in the mixer, the PEQ-2 gives you much more precise control over the tone color. The device consists of two independent, fully parametric equalizers and is most often used as an insert effect, in mono or stereo.

About the two EQ modules

The two independent EQs are labeled “A” and “B”.

- **EQ A is always active (provided that the effect device is in “On” mode and that you have set the Gain to a value other than 0).**
- **To activate EQ B, click the button next to the EQ B parameters, so that the LED lights up.**

► **If you only use one EQ, it’s a good idea to turn EQ B off, to conserve computer power.**

Parameters

For both EQs (A and B), the following parameters are available:

Parameter	Description
Frequency	This determines the center frequency of the EQ, e.g. at which frequency the level should be decreased or increased. The range is 31 Hz to 16 kHz.
Q	This governs the width of the affected area around the set center frequency. The higher the value, the narrower the affected frequency range.
Gain	Specifies how much the level of the selected frequency range should be boosted (positive values) or lowered (negative values). The gain range is ±18 dB.

About the graphic display

The graphic display to the left in the device panel shows the frequency response curve as set by the EQ parameters. This gives a visual feedback and helps you tailor the EQ settings.

CV Inputs

The following CV inputs are available on the back panel of the device:

- **Freq 1 CV.**
 Allows you to control the frequency of EQ A from another device, creating subtle or dramatic EQ sweeps depending on the Q and Gain settings.
- **Freq 2 CV.**
 Allows you to control the frequency of EQ B in the same way.

Chapter 46
The Combinator



Introduction



A custom made Combinator front panel.

The Combinator is a special device that allows you to save and recall any combination of Reason/Rack Extension/VST (Reason stand-alone only) devices (instruments, effects, mixers etc.) and their internal connections. A saved Combinator setup can be loaded as a patch, called a "Combi". The Combinator device itself acts as a container for the devices in a Combi.

The basic idea behind the Combinator device is simple, but very powerful. Being able to save multiple devices as a Combi enables you to instantly recall any type of setup, however complex, as simply as loading a patch!

The original version of the Combinator was first released with Reason 3. In Reason 12 we have made a major overhaul, to make it even more powerful and flexible!

Here are some of the Combinator features:

- **Design your own custom front panel with back-drops, control configurations and layouts.**
 Now you can choose what controls - and how many of each - you want to have on the Combinator front panel. You can also select the height of the Combinator panel (1U to 6U) and add your custom back-drop image to use as background on your Combinator.
- **Create split or layered multi-instruments.**
 Add any number of instrument devices and play them as a single layered instrument. Instrument devices in a Combi can also be assigned to specific keyboard/velocity zones.
- **Save instrument/effect combinations.**
 Save an instrument together with your favorite effect(s).
- **Create multi-effect devices.**
 You can create and save complex effect chains as Combis.

About the Combi patch format

The Combinator saves files in the Combi (.cmb) patch format. When you load a ready-made Combi patch, all devices included in the Combi, their corresponding parameter settings and audio and CV connections are instantly recalled.

The Factory Soundbank includes many preset Combinator patches, divided into various categories.

There are two basic types of Combi patches; instrument and effect combis.

- **Instrument Combis contain instrument devices and generate sound.**
- **Effect Combis contain effect devices and are used to process sound.**

Combinator overview

In the picture below an unfolded Combinator device is shown with its Editor and Devices container. The Combinator in this example contains a pre-made patch, that's why there are devices in it, and assigned controls.



The front of the Combinator consists of the following elements (from the top down):

- **1. The narrow panel at the top is always shown, even when the whole Combinator is folded.**
 It contains a display which (amongst other things) shows the name of the currently loaded Combi, and standard Select/Browse/Save patch buttons.
- **2. Next is the Control panel, which is always shown if the Combinator is unfolded.**
 You can configure and design your own unique control panel, see "Configuring the Combinator panel".
- **3. The Editor contains views for mapping devices and controls, and for configuring the Control panel.**
 Here are also parameters for Key and Velocity Zone mapping as, well as Modulation Routing settings.

 The Editor can be shown/hidden by clicking the "Editor" button on the Control panel.
- **4. At the bottom are the devices included in the Combi.**
 At the top of the device section is an 8-channel stereo summing mixer, which allows for quick audio routing of devices in the Combi. The summing mixer is always present and cannot be removed.

 Devices can be shown or hidden by clicking the "Devices" button on the Control panel.

Creating a Combinator device

Creating an empty Combinator device

→ **Add a Combinator device from the Utilities palette in the Browser.**
Either double click the device or drag and drop from the device palette to the rack.

→ **Select "Combinator" from the "Utilities" sub-menu on the Create menu.**
This will create an empty Combinator. Empty Combinator devices can be used as a starting point when creating new Combi patches. It also allows you to browse for existing Combi patches.

Creating a Combinator by combining devices

You can create a Combinator device by combining existing devices:

1. **Select two or more devices in the rack by [Ctrl]-clicking (Win) or [Shift]-clicking (Mac).**
The devices do not have to be adjacent in the rack.

! **Note that you should never select any Mix Channel or Audio Track devices (Reason stand-alone only), since this will disable the "Combine" function described below.**

2. **Select "Combine" from the Edit menu or context menu.**

This creates a Combinator device containing the devices that were selected according to the following rules:

- **A sequencer track is created for the Combinator, just as when creating instrument or pattern devices.**
- **The new Combinator device appears at the position below the selected device that is farthest down in the rack.**
- **The selected devices are moved to be contained within the Combinator's "container".**
Their internal order is not changed. Reason attempts to autoroute the first "input device" and first "output device" to the Combinator Mixer To/From Devices connectors - see "About internal and external audio connections". Other connections are unchanged.
- **Devices outside the Combi remain in the same order as before.**

! **Please see "Adding devices to a Combi" for details about auto-routing.**

Creating a Combinator by browsing patches

Double clicking, or dragging & dropping a Combinator (.cmb) patch from the Browser to the rack will automatically create the Combinator device in the rack and load the selected patch.

About internal and external audio connections

Unlike other devices, the Combinator contains both external and internal audio connections.

- **External connections are used to connect a Combinator to devices outside the Combi.**
- **Internal connections are how devices within the Combi are connected.**

External audio connections



- **The Combinator Input connectors are the inputs to the Combinator (used for effect Combis only).**
 Internally, these inputs are connected to the "To Devices" connectors of the Combinator Mixer (see below).
- **The Combinator Output connectors carry the audio output of the Combinator Mixer.**
 This output connects with a device outside the Combi, normally a mixer. Internally, this output is connected to the "From Devices" connectors. When you create a new Combinator this output will be auto-routed to the first available input channel in the I/O device.

Internal audio connections



- **The "To Devices L/R" outputs on the Combinator Mixer connect to an input on a device in the Combi when the Combinator is used as an effect.**
- **The "From Devices" inputs on the Combinator Mixer is where the outputs from devices in the Combi are connected.**
 The "From Devices" outputs consist of 8 stereo output pairs.

About External Routing

If an individual device in the Combi is directly connected to a device outside the Combinator - or if an external device is connected to any of the "From Devices" connectors on the Combinator Mixer - this is known as an "External Routing" connection, which should generally be avoided. The reason for this is that such connections are not saved with the Combi patch.

! **Therefore, all connections to/from a Combi should pass via the To/From Device connectors on the Combinator device if you wish to make the Combi "self-contained".**

If a Combi contains any external routing connections, this is indicated by the "External Routing" LED on the front and back of the (unfolded) Combinator Mixer:



It is still possible to use a Combi with External Routing connections within the context of a song (where all routings are saved with the song). Just keep in mind that the external routing connections will not be part of the Combi patch!

How to avoid external routing connections

As explained previously, all connections to/from a Combinator device should pass via the To/From Device connectors in order to make the Combi self-contained. If you should have used up all "From Devices" connectors you need to include a mixer device and route the outputs of that mixer to a free "From Devices" connector pair.

► **If you should run out of "From Devices" connectors, or want more mixing capabilities, like panning and Effect Sends, the Line Mixer 6:2 is ideally suited for mixing device outputs in Combis (see "The Line Mixer 6:2").**

Adding devices to a Combi

Creating new devices in a Combi

To add devices to a Combi, you can drag and drop them from the device palette in the Browser as usual.

If you want to add devices using the Create menu or by double clicking in the device palette, you need to make the Combinator the target device in one of the following ways:

→ **Click in the empty area at the bottom of the open device container so that the red insertion line is shown:**



If you create a device with the insertion line showing, the new device will appear at the bottom of the Combi container.

→ **If the Combi already has devices in it, select a device in the Combi (but not the Combinator device itself).** When you add a device from the Create menu it will appear below the selected device (just like in the rack).

! **No sequencer tracks will be automatically created for devices added to a Combi.**

About auto-routing

The auto-routing of devices in a Combi is similar to devices in the rack:

- **If a device in a Combi is selected, the new created device will appear below the selected device according to standard rules.**
- **If an effect is selected and you create a new effect device, these will be connected serially.**
- **If an instrument device is selected and you create an effect it will be connected as an insert effect to the instrument device.**
- **If a mixer is selected and you create an effect it will be connected as a mixer send effect.**
- **If an instrument device is selected and you create another instrument device it will be added below the selected device and connected to the first available Combinator Mixer input channel.**
- **If you hold down [Shift] and create a new device, no auto-routing will take place.**
- **If you hold down [Option] (Mac) or [Alt] (Windows) and create a new device, a sequencer track will be created for the device.**

→ **If you add a device to an empty Combi, its output(s) will be auto-routed to the first available "From Devices" connector(s) of the Combinator Mixer. For effect devices, the input will also auto-route to the "To Devices" connectors.**

Adding existing devices to the Combi from the rack

You can move devices in the rack that are currently outside the Combi into the Combinator holder. This works as follows:

1. **If you want to add more than one device at the same time, [Ctrl]-select (Win) or [Shift]-select (Mac) the devices.**
2. **Click on the panel area of a device.**
 You can click anywhere outside the actual parameters and displays.
3. **Drag the device(s) into the Combinator.**
 A thick orange vertical line indicates where the device(s) will be positioned. Note that the orange line can be to the left of a device (indicating that the moved device will be inserted before the other device) or to the right of a device (indicating that the moved device will be inserted after the other device). You can also add devices to the empty space at the bottom of the Combi holder.

- **No auto-routing will take place.**
 The existing connections to the rack will be kept, so there may be external routing connections (see "About External Routing"). If there are, "External Routing" will be shown the Combinator name display (and on the back of the Combinator a LED indicator will be lit).
- **If you drag devices into the Combi with [Shift] pressed, the devices will be disconnected from the rack and auto-routed according to the insert position.**
 The routing between dragged devices will be kept. If it isn't possible to auto-route a dragged device (e.g. if there are no free From Devices inputs on the Combinator Mixer), the device will become unconnected.
- **If you drag devices into the Combi with [Alt](Win) or [Option](Mac) pressed, the devices will be copied. No auto-routing takes place.**
 Using [Shift] at the same time will attempt to auto-route according to the same rules as described above.

Combining two Combis

- **If you select two Combinators in the rack and then select "Combine" from the context menu the lower Combi will be uncombined and the devices added to the upper Combi in the rack when combining.**
 Existing routing will be unchanged.

Combinator handling

Moving the entire Combi

This works much the same as for other devices in the rack.

- **Select the Combinator by clicking on the Control panel background and dragging to a new position.**
 An outline of the Combinator is shown when you drag, and an orange line shows the insert position. All connections are kept.

Moving devices within a Combi

Just as for devices in the rack, an outline of the devices is shown when you drag, and an orange line shows the insert position. All connections are kept.

Moving devices out of a Combi

If you move devices out of a Combi the following applies:

- **The routing in the Combinator is unchanged, and the External Routing indicator is likely to light up.**
 If you press [Shift] when dragging, the device(s) will attempt to auto-route to the rack according to standard rules.

Deleting devices in a Combi

This works exactly as for devices in the rack.

- **Select the device and then either select "Delete Devices and Tracks" from the Edit menu or context menu, or press [Backspace].**

Uncombining devices

You can uncombine a whole Combi or selected devices within a Combi in the following way:

- **If you select a Combinator and then select "Uncombine" from the Edit/context menu, the Combinator device will be removed, and all devices contained in the Combi will be moved to the rack.**
 The devices previously connected to the To/From Devices connectors will now be connected to the rack in the same way the Combinator device was (via the Combinator output and input).
- **If you uncombine a few selected devices in a Combi these will be removed from the Combi and added to the rack below the Combi.**
 Connections are unchanged, so external routing is likely to happen.



Configuring the Combinator panel



This is the default Combinator panel - the one that appears after you have selected "Reset Device" from the context menu. Like standard instrument devices it features Pitch and Mod wheels and various controls.

About the virtual controls

- **The default four Rotary knobs and buttons in the middle of the Control panel are "virtual" controls that can be assigned to parameters and functions in devices contained in the Combi.**
 These controls are by default not assigned to any parameters in new Combis.
- **You can also add or remove controls, to design your own custom panel, see "Adding and positioning controls on the front panel".**
- **You assign parameters to the controls in the Modulation Routing section of the Editor panel (see "Using Modulation Routing").**
- **Movements of the virtual controls can be recorded as automation.**
- **Each control can be assigned to any number of parameters.**
- **Up to four virtual controls can be modulated from the Control CV inputs on the rear panel, see "Control CV In".**
- **Double clicking on the label for a control lets you type in an appropriate name for it.**

The Pitch Bend and Modulation wheels

The Pitch and Mod wheels on the Control panel will mirror the corresponding actions on your master keyboard, just like for standard instrument devices.

- **When a Combinator device has Master Keyboard input and the Combi contains several instrument devices these will all receive pitch bend and modulation data.**
 The settings in the instrument devices determines what happens when applying pitch bend or modulation.
- **For example, moving the Mod wheel could apply vibrato for one device and change the filter cutoff frequency for another device.**
- **Pitch Bend will also be applied according to individual settings in the Range field for all instrument devices in a Combi.**

Opening the Configuration panel

1. **Open the Editor by clicking the Editor button at the top of the Combinator front panel:**

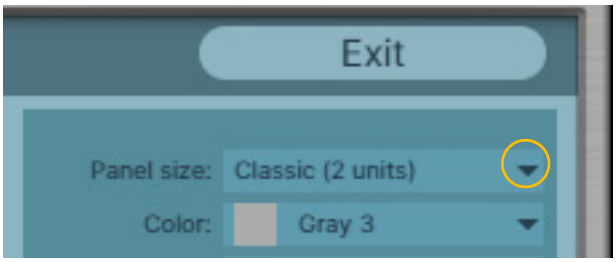


2. **Click the Configure button to open the panel Configuration panel:**



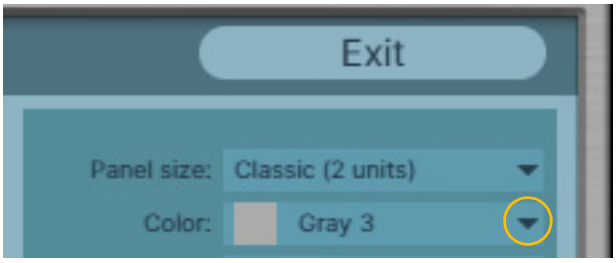
Setting the front panel size

→ **Select the desired panel height from the “Panel size” drop-down list.**
You can choose from 1U (rack unit) to 6U.



Selecting front panel background color

→ **Select the desired panel background color from the Background Color drop-down list.**

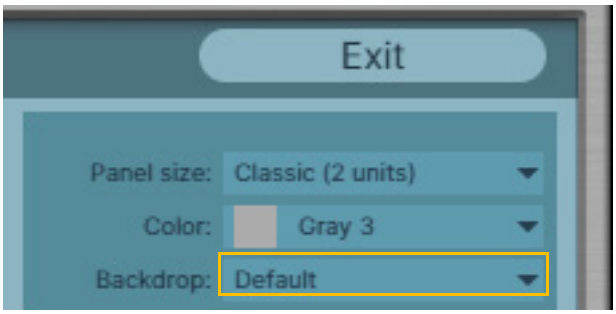


Selecting a backdrop image



This function allows you to change the “skin” of the Combinator front panel. You can design your own labels for the assignable controls, and change the color and look of the whole panel.

→ **Click the Backdrop drop-down on the Config panel in the Editor:**



The Backdrop drop-down features the items “None”, “Default” and “Load...”

- **“None” leaves the panel with the selected background Color (see “Selecting front panel background color”).**
- **“Default” is a semi-transparent texture which gives a higher quality look to the front panel.**
 Since the texture is semi-transparent the selected background Color will shine through.
- **Selecting “Load...” will open the image browser.**
 Here you can select image file in JPEG (.jpg) or PNG format.

 Instead of selecting “Load...” from the drop-down you could select the Combinator device and choose “Select Backdrop...” from the Edit/context menu.

 ► **Transparent PNG files are also supported, if you want to layer a backdrop image on top of a background color.**
- **For full hi-res support the width of the image should be 3770 pixels.**
 The height depends on what front panel size you have selected, see “Setting the front panel size”.

 These are the optimal height dimensions for each of the front panel sizes:

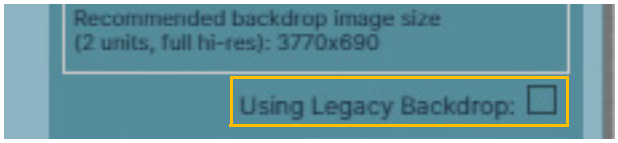
 1U: 345 pixels
 2U: 690 pixels
 3U: 1035 pixels
 4U: 1380 pixels
 5U: 1725 pixels
 6U: 2070 pixels

 If you select an image with other pixel dimensions, the image will be automatically scaled.
- **The knobs, buttons, patch name display and patch buttons cannot be redesigned using a backdrop.**

→ **If you wish to design your own text labels for the virtual controls, you should first remove the original text labels:**
Double click on a control label, remove the current text and press [Enter]. Alternatively, uncheck “Show Label” in the Configuration panel in the Editor, see “Changing the appearance of the controls”).

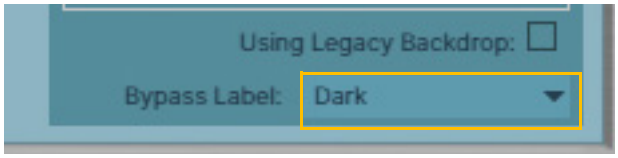
→ **To remove a Backdrop, select “Remove Backdrop” from the Combinator context menu - or select “None” in the Backdrop drop-down on the Config panel in the Editor.**
The currently selected Background Color of the Combinator panel is restored.

About “Using Legacy Backdrop”



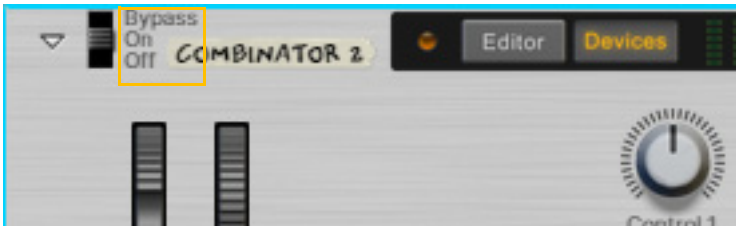
This is automatically turned on when you load a legacy Combi patch with a backdrop, and will show buttons in the same way as on the original Combi (including Show Editor/Show Devices buttons on the panel) and hide the Bypass/On/Off label as this is expected to be in the backdrop.

About the “Bypass Label”



The “Bypass Label” item is available only when the “Using Legacy Backdrop” is de-selected.

→ **Select the desired color (or presence) of the Bypass/On/Off text from the drop-down.**
The setting affects the color of the text on the top left of the Combinator panel:



About the “Combi Layout Templates” folder

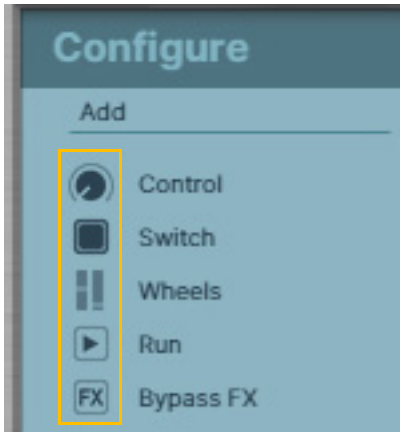
In Factory Sound Bank/Combinator Patches/ there is a folder named “Combi Layout Templates”. This folder contains a selection of pre-made Combinator panel configurations and layouts that can serve as starting points/inspiration for your own Combinator designs.

Selecting, positioning and editing front panel controls

1. **Open the Editor by clicking the Editor button at the top of the Combinator front panel.**
2. **Click the Configure button to open the panel Configuration panel.**

Adding and positioning controls on the front panel

To the left on the Editor Configuration panel are a selection of pre-made knobs, faders, buttons and wheels that can be freely laid out on the Combinator front panel:



Controls selector on the Configuration panel.

1. **Click a desired control icon on the Configuration panel.**
 The control shows up with a frame around it on the Combinator front panel.

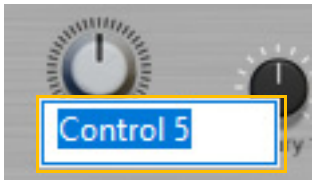


Adding a new control to the front panel.

2. **Click and drag the control to the desired position on the Combinator front panel.**
 Visual guides appear as you move the control so you can easily place the control aligned with other controls already on the panel.

→ **Use the [Arrow] keys on the computer keyboards to adjust the position, if desired.**

→ **Double click the label and type in a new name, if desired:**



3. **Click a new control to add another one.**

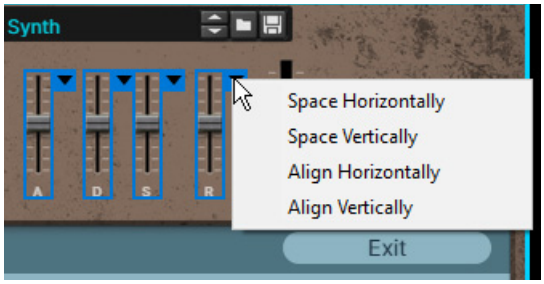
→ **You can also select multiple controls, by dragging or by [Shift]-clicking on the panel, and move them together on the front panel.**
 Alternatively, [Command]/[Ctrl]-click the controls in the Panel Elements list to select multiple controls.



! **The maximum number of controls that can be placed on a Combinator panel is 32 knobs/faders plus 32 buttons, 2 wheels (pitch and modwheel), 1 Run button and 1 Bypass FX button.**

It's also possible to align and distribute a selection of controls, to get perfect position with even spacing on the panel.

→ **Select the group of controls you want to align/distribute, click the black arrow button on the control you want to use as reference, and select action from the pop-up:**



About the "Run" and "Bypass FX" buttons



The "Run" button can be used to start/stop all (Reason) pattern devices included in the Combi. This works exactly as pressing the Run button on the pattern device panel. The on/off status of this button is not saved with the Combi patch. Pressing play on the transport panel in the sequencer will automatically activate "Run Pattern Devices".

→ **In the Reason Factory Sound Bank, Combi patches containing pattern devices have "(run)" at the end of their patch names.**

The "Bypass FX" button allows you bypass all effect devices in a Combi. It works as follows:

- **All insert effect devices in the Combi are switched to Bypass mode.**
- **All effects connected as send effects to a mixer device are switched off.**
- **Clicking this button will not affect effect devices that were bypassed or turned off already.**

Changing the appearance of the controls

It's possible to change the appearance of the controls.

1. **Click the control you want to edit, either on the Combinator front panel or in the Panel Element list on the Configuration panel.**
 The characteristics of the selected control is now displayed in the Settings section:

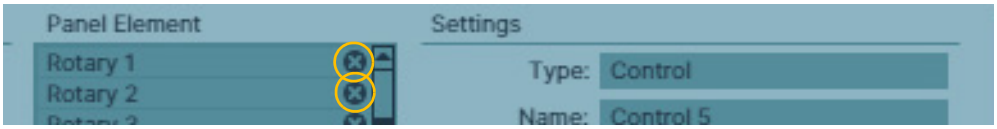


- **The control type is shown in the Type display.**
- → **The Name display shows the control name.**
 (To edit the name, double click the name label on the Combinator panel.)
- → **Select control style from the Style drop-down list.**
- → **Select the Scale appearance (if the control features a scale)**
- → **Click the check mark to show/hide the control name label on the Combinator front panel.**
- → **Select the label text color from the Text Color drop-down list.**
- → **Choose where you want the text label positioned around the control from the Position drop-down list.**

Deleting controls on the front panel

1. **Select a control (or multiple controls, by dragging or by [Shift]-clicking the controls on the front panel).**
 Alternatively, click in the Panel Element list ([Command]/[Ctrl]-click to select multiple controls).
2. **Press [Delete]/[Backspace] on the computer keyboard to delete the selected control(s)**

→ **Alternatively, click the X next to the control name in the Panel Elements list to delete one control at a time:**



Assigning panel controls to parameters in the Editor

The Editor is used for key mapping and setting velocity ranges for instrument devices, and for Modulation Routing; assigning device parameters to the knobs/buttons/sliders on the Control panel.

➔ **To bring up the Editor panel, click the "Editor" button on the unfolded Control panel.**
The Editor appears below the Control panel.

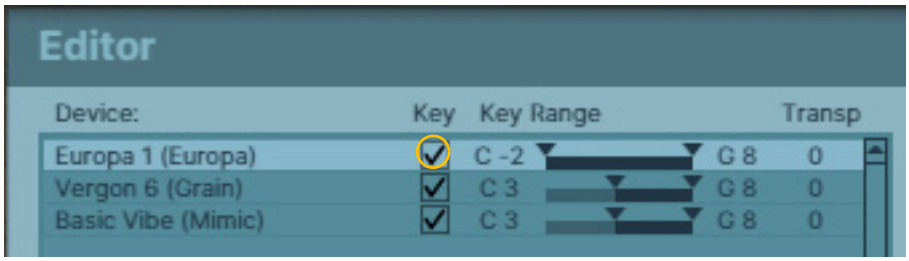


A fully assigned Combinator.

- **To the left on the Editor panel the devices in the current Combi are listed in the order they appear in the Combinator container - from top to bottom.**
 Clicking on a device in the list selects it for editing.
- **To the right is the Modulation Routing section where you can assign parameters to the controls on the Control panel.**
 See "Using Modulation Routing".

Key Mapping instrument devices

About the Key (receive MIDI Notes) column



To the right of the Device name is a Key column.

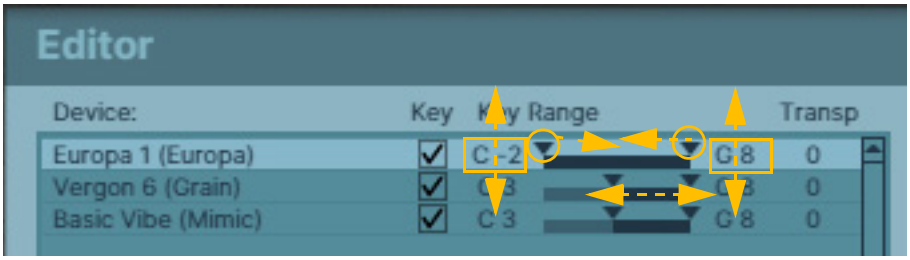
➔ **Click to select if the device should respond to MIDI Note input.**

- **If you deactivate the "Key" checkbox the selected device will not respond to incoming MIDI note messages.**
 For devices that cannot use MIDI notes (such as most effect devices), there is no checkbox in this column.

Setting the Key Range

Each instrument device can have its own separate key range, the lowest and the highest key that will trigger the device. This allows you to create splits and layers for instrument devices in a Combi.

1. **Make sure the "Key" checkbox is ticked for the instrument devices you want to edit the Key Range.**
 The Key Range is shown as a horizontal bar, and as note numbers on either sides of the bar. By default, the entire note range is selected (C -2 to G 8).



There are several ways you can change the Key Range:

- **By clicking in the Key Range value fields and moving the mouse up or down.**
- **By moving the handles of the horizontal bar.**
- **By dragging the horizontal bar itself you can also move entire key zones horizontally, thereby changing their key ranges.**

2. **Using either method, set the desired key range for the desired device(s).**
 When done, the device(s) will only play back notes in the set key range.

- **By setting up key ranges for devices in a Combi, you can create split instruments.**
 For example playing notes below C 2 could trigger a device playing a bass sound, whereas playing notes above C 2 could trigger a device playing a pad sound.
- **Instrument devices in a Combi that share the same key range will be layered - i.e. play at the same time.**
 This given that no velocity ranges have been set up - see below.
- **You can of course set up overlapping ranges where notes within a set key range will layer two (or more) devices, but notes above and below the set range will play separate devices.**

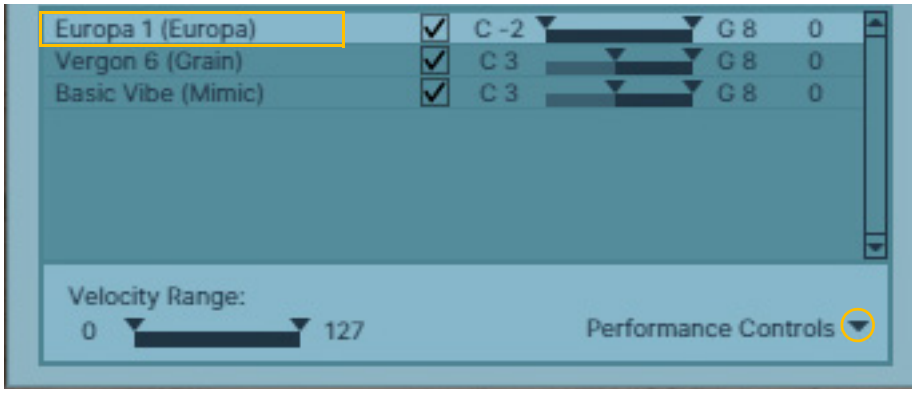
About the Transpose function



The Transpose field to the far right on the panel allows you to transpose the instrument devices. It will not shift the key mapping, just the pitch of the device.

- **Click and drag up/down in the Transpose field to edit the transposition.**
 The range is +/- 3 octaves, in semitone steps (+/-36 semitones).

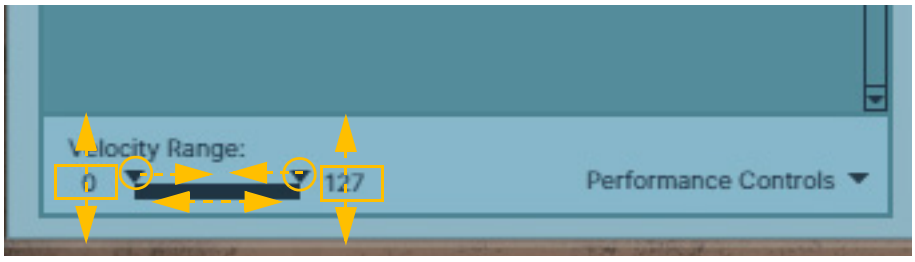
Enabling/disabling Performance Controls



To the bottom right of the panel is a drop-down list that features all standard MIDI Performance Controllers (Pitch Bend/Mod Wheel/Breath/Expression Pedal/Sustain Pedal/Aftertouch) for the selected instrument.

- **The check boxes in the drop-down list allow you to control whether MIDI Performance Controller data is to be received for each instrument device in a Combi.**
- **If you deactivate any of the Performance Controllers, the corresponding controller(s) will not be received by the selected instrument device.**
 All Performance Controllers are on by default.

Setting Velocity Ranges for instrument devices



When instrument devices are set up so that their key ranges overlap – completely or partially – you can use velocity switching to determine which devices should be played back depending on how hard or soft you play on your MIDI keyboard.

This is done by setting up velocity ranges.

Each time you press a key on your MIDI keyboard, a velocity value between 1-127 is sent to Reason. If you press the key softly, a low velocity value is sent and if you press it hard, a high velocity value is sent.

This velocity value determines which devices will be played and which will not.

1. **Select an instrument device in the Device list (non-instrument devices, e.g. effects and mixers do not have velocity ranges).**
 By default, the entire range is selected (0 - 127).

There are several ways you can change the current Velocity Range:

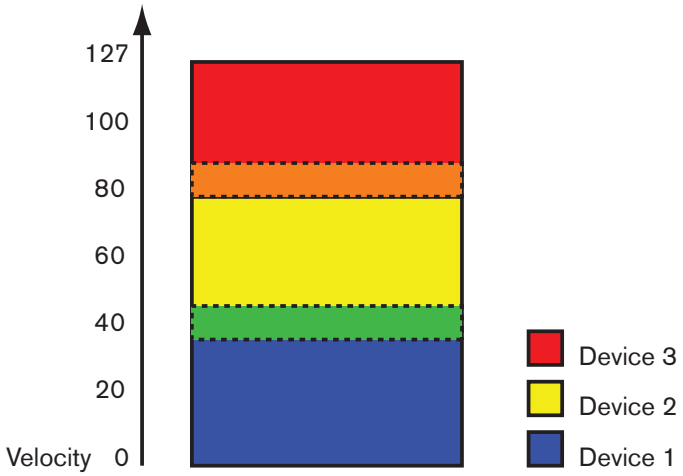
- **By clicking in the "Lo" and "Hi" value fields and moving the mouse up or down.**
- **By moving the handles of the horizontal bar.**
- **By dragging the horizontal bar itself you can also move entire Velocity Range.**

2. **When you have set a range, the device will only be triggered by notes played within this velocity range.**

About overlapping Velocity Ranges

You can set overlapping velocity ranges. Here's an example of how this can work:

- **Device 1 has a velocity range from 1-45.**
- **Device 2 has a velocity range of 33-88.**
- **Device 3 has a velocity range of 76-127.**



Now, velocity values between 33 and 45 will trigger notes from both Device 1 and Device 2. Likewise, velocity values between 76 and 88 will trigger sounds from Device 2 and Device 3.

Using Modulation Routing



The Modulation Routing section to the right on the Editor panel allows you to assign any parameter or function in devices included in a Combi to any of the virtual controls on the Control panel.

You can also control Combi parameters by connecting external CV modulation sources to any of the Source CV inputs on the back of the unfolded Editor panel, see "Wheel CV In and Source CV In".

About Rotary/Slider and Switch controls

The virtual Rotary/Slider and Switch controls operate much like the equivalent controls on the real devices:

- **A Rotary/Slider control can either smoothly change parameter values (e.g. a level control), or step through fixed values (like the Oscillator waveform spin controls on a Subtractor).**
- **A Switch control will switch between two set values like an on/off switch.**
 Worth noting is that there are buttons on several Reason devices that will step through a series of values, for example LFO Waveform buttons. If LFO Waveform is assigned to one of the virtual Buttons you will only be able to switch between two of the six LFO waveforms (which waveforms is determined by the Min/Max range).
- **The available range for each selected parameter is shown in the Modulation Routing Min/Max fields.**
 Most sliders and rotary knobs on the actual devices have the standard 0-127 or -64 to 63 range. Selectors and spin controls can have any value range.

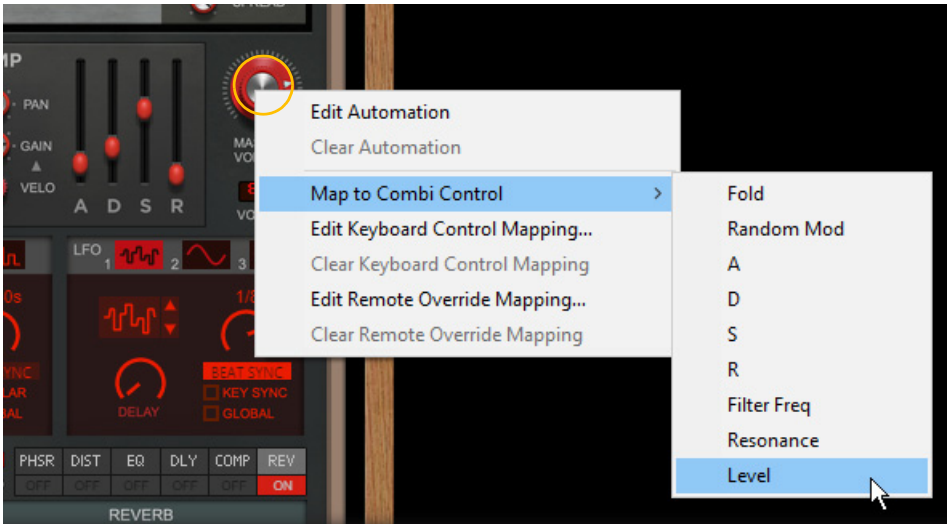
About the CV Inputs

On the back of the Editor panel are CV Modulation Inputs for connecting external CV signals to modulate any of the parameters that are accessible in the Target section in the Editor, see "Wheel CV In and Source CV In".

Assigning panel controls to device parameters

Assigning Combinator panel controls to device parameters can be done in two different ways:

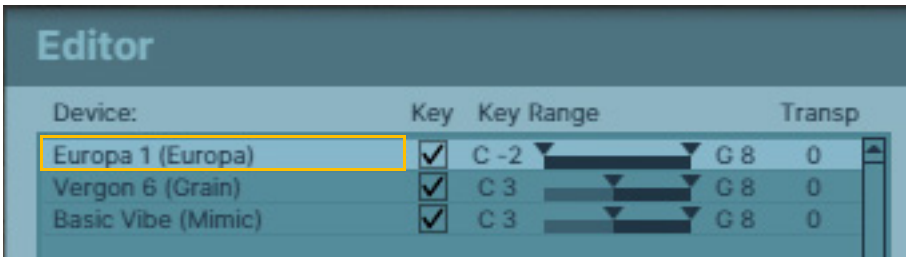
1. **Right click the desired device parameter and select the desired Combinator panel control to assign it to from the "Map to Combi Control" pop-up:**



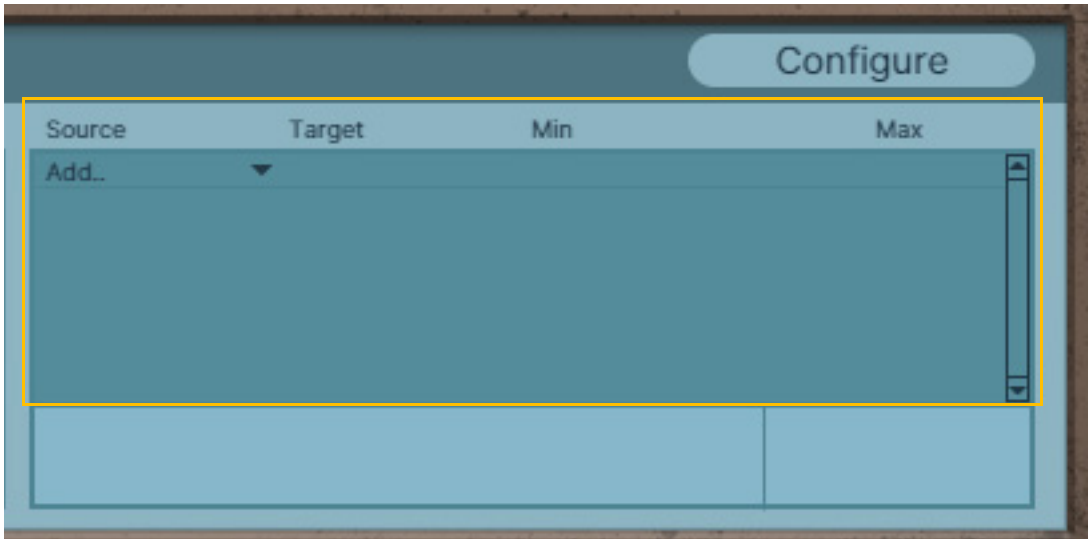
Assigning the "Master Volume" control on an Europa device to the "Level" control on the Combinator panel.

If you chose the method described above you can continue directly to step 4 below in this description.

→ **Alternatively, select the device you wish to assign parameters on in the device list to the left:**

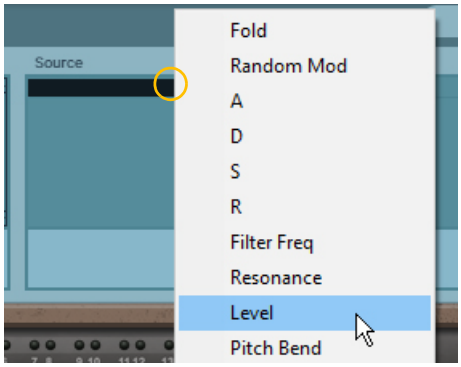


The Modulation Routing section to the right contains four columns:



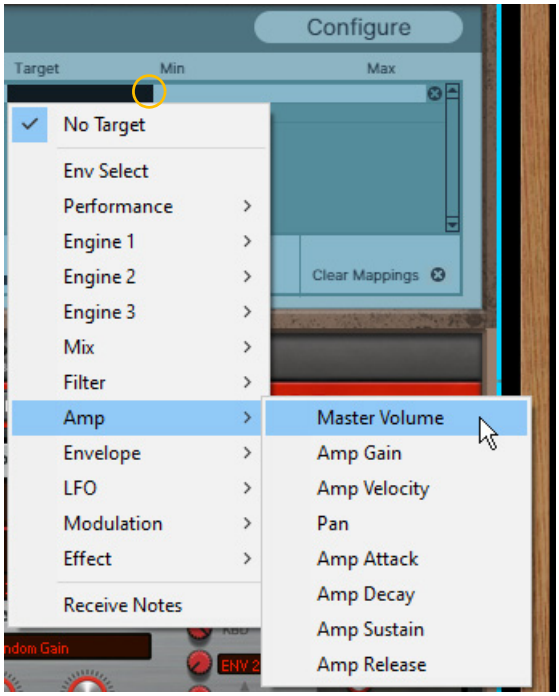
- **In the Source column, any currently assigned Combinator panel controls are listed.**
 Each field can be changed to any Control/Slider/Switch/CV Input or Performance Controller by clicking the arrow and selecting from the drop-down list.
- **The drop-down lists in the Target column contain all parameters for the currently selected device.**
 Lastly in each Target drop-down list is the option to receive note data or not. This is useful if you want to temporarily enable/disable MIDI Note input for specific devices, e.g. to temporarily enable layered instruments - or switch instrument playback.
- **The Min/Max columns allow you to specify a value range for the virtual control.**
 It's also possible to set the Min value higher than the Max value, meaning that the value will decrease when you turn up the knob/slider.

2. **Click the small black triangle in the Source column and select the Combinator panel control from the drop-down list:**



Setting the "Level" Combinator panel control as Source.

3. **Click the small black triangle in the Target column and select the desired Target parameter from the drop-down list:**
In the drop-down, all the available parameters for the device are listed.

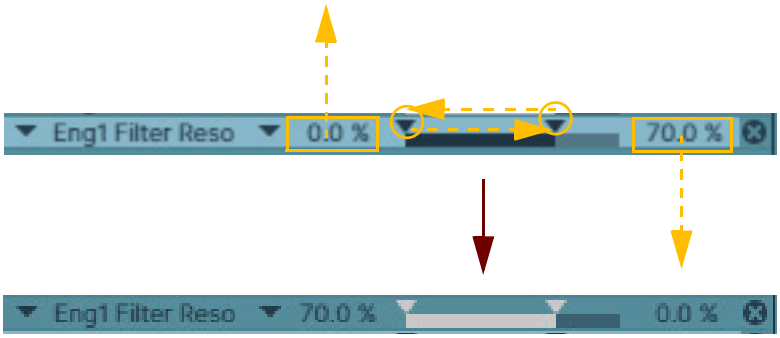


Assigning the Europa "Master Volume" parameter as Target.

4. **You can specify a range for the assigned parameter by clicking in the Min and Max columns and moving the mouse up or down, or by dragging the handles of the horizontal Range bar, or by dragging the Range bar (to offset the range).**
By the maximum available range is set by default:

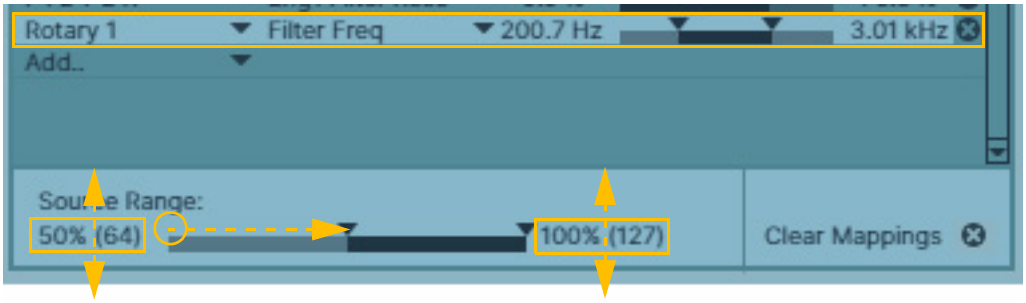


- **It's also possible to "reverse" the range by dragging the Range handles past each other, or dragging the Min/Max boxes so that the minimum value becomes max and the maximum value becomes minimum.**
 The result is that the Target parameter value will decrease when you turn up the Source knob/slider, and vice versa. Here is an example of how to reverse the range for a parameter:

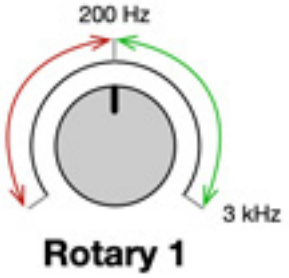


The Filter Reso parameter edited to have a reversed Range, as indicated by the light gray Range bar.

- **It's also possible to set a Source Range for each Source control, so that the knob/slider only affects the Target parameter in a certain region/interval of the Source control:**



The Filter Freq stays at 200 Hz in this span.

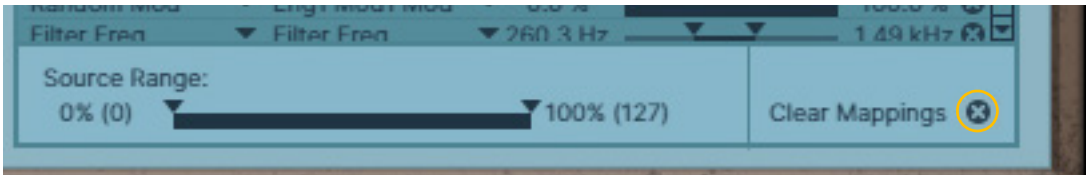


Linear change from 200 Hz to 3 kHz in this span.

Source Range for Rotary 1 set to 50-100%.

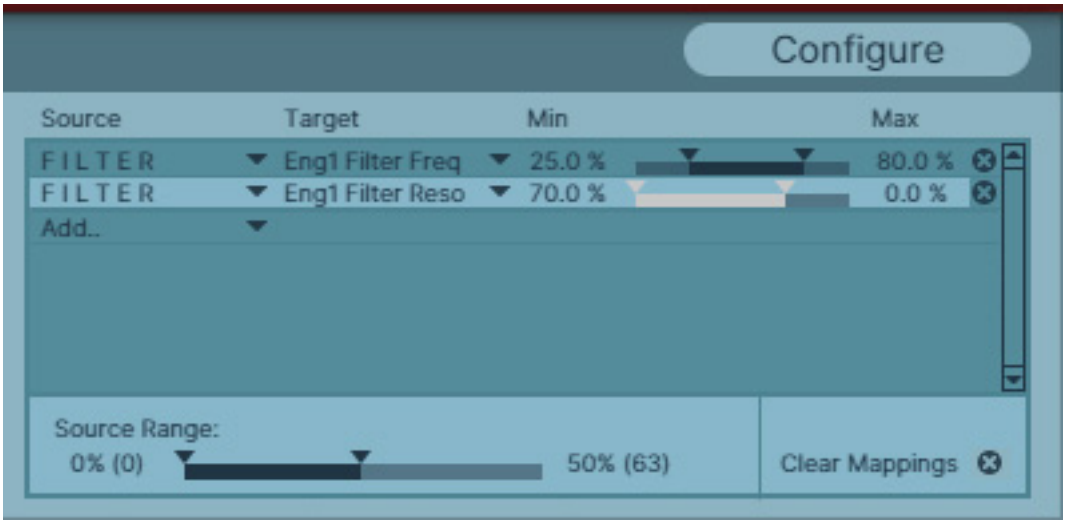
5. **If you select another device in the Device list to the left in the Editor, you can assign another parameter to the same Rotary or Button control using the same basic methods.**
 This means that you can create multi-function controls that operate simultaneously on several parameters. E.g. if you have two Subtractors and a Malström in a Combi you could create a "master" filter cutoff knob, that controls this parameter for all three devices. See "Controlling multiple Target parameters differently from a single Source control" for an example.

→ **If you want to clear all mappings and start from a clean slate, click the "Clear Mappings" X button:**



Controlling multiple Target parameters differently from a single Source control

You can also have the same Source control assigned to multiple Target parameters on the same device. In the example below the “Eng1 Filter Freq” and the “Eng1 Filter Reso” parameters on the Spectral Filter in a Europa device are assigned to the same FILTER control on the Combinator panel:



The “Eng1 Filter Reso” parameter Range is reduced and reversed (light gray Range bar), so when you turn up the FILTER knob on the Combinator panel, the Filter Frequency is raised but the Resonance is decreased. The “Eng1 Filter Reso” parameter also has the Source Range set to 0-50%, which means the Resonance will decrease only during the first half of the FILTER knob range.

The picture below shows how the “Eng1 Filter Freq” and “Eng1 Filter Reso” parameters are affected when you turn up the FILTER knob on the Combinator panel. You can also see the differences in the Spectral Filter display:



Controlling both the Freq and Reso parameters from the FILTER control on the Combinator panel.

Controlling a single Target parameter from multiple Source controls

This is also possible, by simply selecting the same Target parameter from different Source controls in the Modulation Routing section.

CV Connections



The rear panel of the Combinator.

CV connections between devices in the Combi are saved with the Combi patches. This is also true for CV connections between devices in Combi and the Combi itself - e.g. if you have connected a Matrix in the Combi to one of the CV inputs on the back of the Combi panel.

The following CV connectors can be found at the back of the Combinator:

Sequencer Control inputs

The Sequencer Control CV and Gate inputs allow you to play the Combinator from another CV/Gate device (typically a Matrix or a RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity.

Control CV In



These CV inputs can be used for controlling four selectable controls on the Combinator front panel.

- **Click the drop-down and select which panel control you want to modulate.**
 All currently available front panel controls are shown in the drop-down list.
- **Attenuate the input signal with the corresponding attenuation knob.**

Wheel CV In and Source CV In

In this section you will find standard CV Pitch Bend and Mod Wheel modulation inputs, as well as eight modulation inputs which can be assigned as Sources in the Editor. Here you can connect external CV modulation sources for modulating any of the Target parameters in the Editor (see "Using Modulation Routing").

→ **Click the Editor button to show the Wheel CV In and Source CV In connectors on the rear panel:**



Next to each CV Input are one sensitivity knob and one polarity switch that work as follows:

- **The sensitivity knobs can be used for attenuating the CV Input signal.**
- **The polarity switches should be used for defining the polarity of the CV Input signal.**
 For example, if you have connected a CV signal from an envelope generator, the switch should be set to Unipolar. If you are modulating from a standard LFO, the polarity switch should be set to Bipolar.



Chapter 47
Pulsar Dual LFO

Introduction



The Pulsar Rack Extension device is a very flexible and versatile dual LFO module. LFOs (Low Frequency Oscillator) are used for generating cyclic modulation signals. A typical example is to have an LFO modulate the pitch of an oscillator to generate vibrato, but there are countless of other applications for LFOs.

Pulsar features two separate LFOs that can be used for modulating parameters in other rack devices. The two LFOs can also modulate each other to generate complex modulation signals. The LFOs in Pulsar can reach way up in the audible frequency range, which opens up for really interesting applications. As an additional feature the LFO rates can also be tracked from a MIDI keyboard.

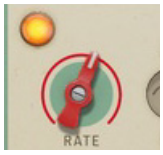
In Reason the Pulsar Rack Extension device can be found on the Utilities palette.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

Panel parameters

LFO 1&2 common parameters

Rate



This controls the LFO rate. The Rate range in Pulsar is very wide and can reach way up in the audio frequency range. The rate is indicated by the lamp to the left above the Rate knob. The rate can also be synced to the sequencer tempo by clicking the Tempo Sync button below the Rate knob (see "Tempo Sync" below). In sync mode, the Rate knob controls the sync resolution.

The LFO 1 Rate can be modulated and/or synced by LFO 2, see "Rate (LFO 2 to LFO 1 Rate)" and "Sync". The Rates can also be modulated from the Envelope, see "Envelope". As a special feature, the Rates can also be controlled from a MIDI keyboard, see "KBD Follow".

Range: 0.06Hz-1.05kHz (synced: 32/4 to 1/64th)

! **When the Rate is modulated, it can reach far beyond the default frequency range.**

Waveform selectors



Here you can select one of nine different LFO waveforms. Besides the standard waveforms (sine, triangle, pulse, etc.) there are random, slope and stepped waveforms. The shape of the waveforms are shown in the display.

! **Note that all waveforms are bipolar, i.e., they generate both positive and negative levels.**

→ **Select waveform for each LFO by clicking the up/down arrow buttons or by clicking and holding in the waveform display and dragging up/down.**

Level

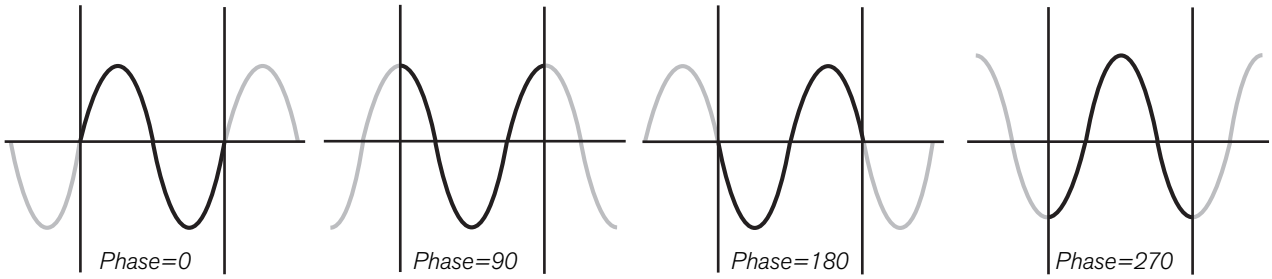


Here you set the output level of the LFO signal. The LFO 1 Level can be modulated by LFO 2, see "Level (LFO 2 to LFO 1 Level)". The Levels can also be modulated by the Envelope, see "Envelope".

Phase



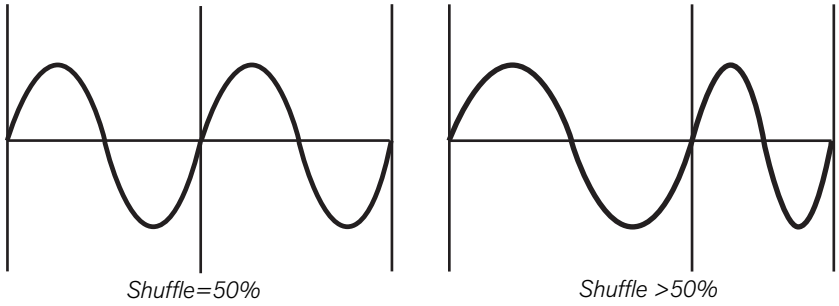
The Phase control lets you offset the phase of the LFO cycle, i.e. decide where in the cycle the waveform should start. The range of the Phase control is 0-360 degrees:



Shuffle



The Shuffle function affects two adjacent LFO cycles, in pairs. Increasing the Shuffle value lengthens the first cycle and shortens the second one:



Note that the total length of the cycle pair is always 2 regular cycles, which means that shuffling will work great also in Tempo Sync mode, see "Tempo Sync".

Range: 50% (no shuffle) to 75%

Lag



The Lag control acts like a lowpass filter on the LFO signal, making the signal smoother. This is especially noticeable on waveforms with sharp edges or transients like the square, sawtooth and stepped waves. On the sinewave you will barely notice any effect since it's already smooth by nature.

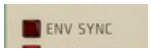
Tempo Sync



Click the Tempo Sync button to sync the LFO Rate to the sequencer tempo. This is great for creating animated effects in sync with the song's tempo. In Tempo Sync mode, the Rate knob controls the sync resolution, see "Rate".

LFO 1 specific parameters

ENV Sync



With ENV Sync active, whatever triggers the Envelope will also restart LFO 1, see "Envelope".

If Tempo Sync is also active (see "Tempo Sync") the LFO will continue to sync to the sequencer tempo. However, *when* you actually press a key or hit the Trig button will affect the LFO cycle start. In some situations this could be perceived as the LFO is losing sync but it's not; it's merely the LFO cycle start that is changing.

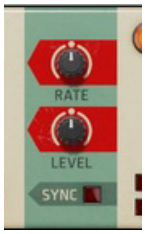
LFO 2 specific parameters

On/Off



Click this to activate LFO 2.

LFO 2 to LFO 1 modulation parameters



Rate (LFO 2 to LFO 1 Rate)

This controls the modulation amount from the LFO 2 signal level to LFO 1's Rate. The result is frequency modulation (FM) of LFO 1.

Level (LFO 2 to LFO 1 Level)

This controls the modulation amount from the LFO 2 signal level to LFO 1's Level. The result is amplitude modulation (AM) of LFO 1.

Sync

With the Sync button on, every new LFO 2 cycle automatically restarts LFO 1.

Envelope



This is an AR (Attack-Release) envelope. The Envelope can be triggered from any of these four sources:

- **The Trig button, see "Trig".**
- **LFO 2, see "LFO 2 Trig".**
- **Envelope Gate In modulation input on the rear panel, see "Envelope Gate In".**
- **MIDI Note On from a connected MIDI keyboard.**

The Envelope can modulate the Levels and Rates of LFO 1 and/or LFO 2.

! **If the envelope is retriggered before all envelope stages are completed, the envelope will simply restart at the current level (similar to how a monophonic synthesizer works).**

LFO 2 Trig

With the LFO 2 Trig button on, the Envelope is automatically synced by the LFO 2 signal. This means that every time LFO 2 begins a new cycle, the envelope is triggered.

Trig

This is a non-latching gate button which gates/triggers the envelope.

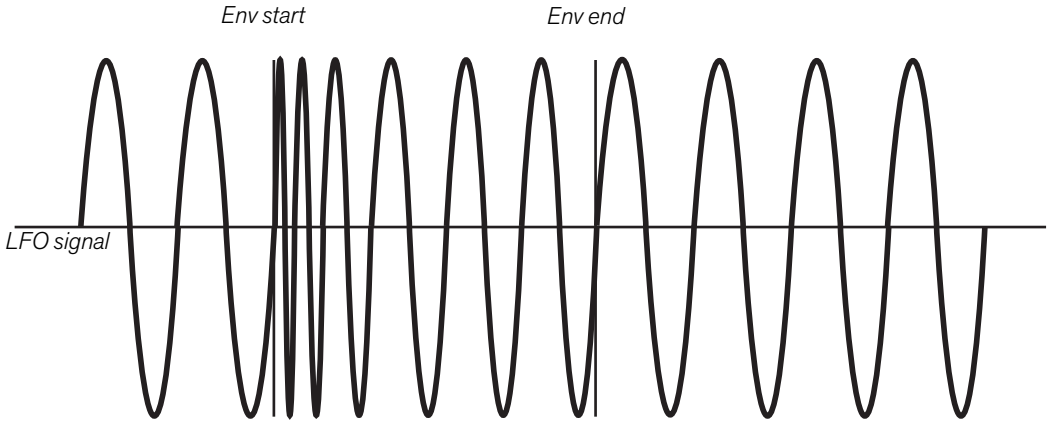
Attack and Release

The attack and release times of the envelope when the gate opens/closes are set with these knobs. For a gated effect set both controls to zero and control the gate time with the Trig button, CV Gate In modulation input or MIDI Notes. You can also let LFO 2 trig/gate the envelope as described above.

Range: 0.1ms-3.00s (Attack) and 0.0ms-10.00s (Release)

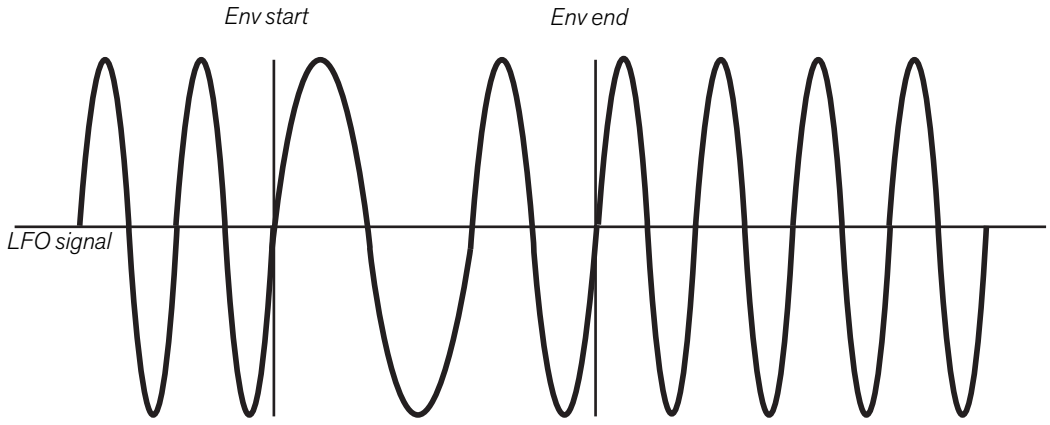
Rate (LFO 1 and 2 Envelope modulation)

The Rate knobs control the modulation amount of the Rate parameters of LFO 1 and 2 respectively. With these you can force the LFO rates to vary according to the envelope's Attack and Release times:



Env Attack = 0, Env Release > 0, Rate modulation = +50%

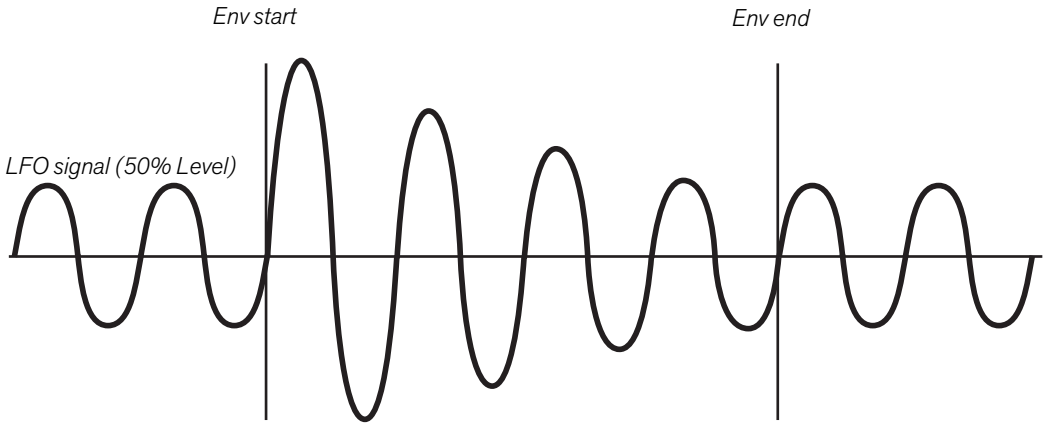
The Rate modulation controls are bipolar, with no modulation at 12 the o'clock position, negative modulation to the left and positive to the right. Negative modulation means that the LFO Rate gets slower during the envelope stage and then goes back to the set Rate:



Env Attack = 0, Env Release > 0, Rate modulation = -50%

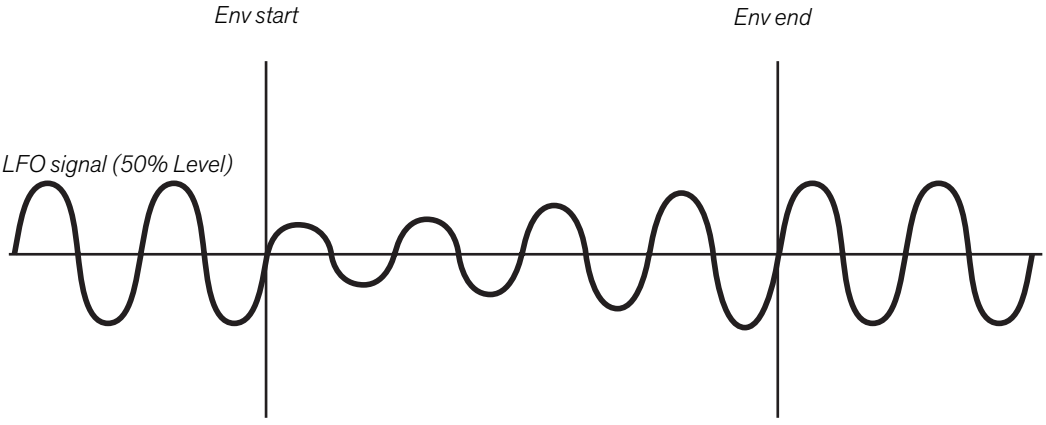
Level (LFO 1 and 2 Envelope modulation)

The Level knobs control the modulation amount of the Level parameters of LFO 1 and 2 respectively:



Env Attack = 0, Env Release > 0, Level modulation = +50%

The Level modulation controls are bipolar, with no envelope modulation at 12 the o'clock position, negative modulation to the left and positive to the right. Negative modulation means that the LFO signal decreases in level during the envelope stages and then goes back to set Level:



Env Attack = 0, Env Release > 0, Level modulation = -50%

KBD Follow



It's possible to control the LFO Rates in Pulsar from a connected MIDI keyboard:

- **Create a sequencer track for Pulsar by selecting "Create Track for Pulsar n" from the context menu (right-click the device in the rack).**
 A sequencer track is created and is automatically selected.

With the KBD Follow knob you can define how the LFO Rates should be modulated by incoming MIDI Note data. The KBD Follow parameter is bipolar:

- **At zero (12 o'clock position), the Rates are not modulated at all.**
- **At 100 the LFO Rates track the incoming MIDI Notes 1:1, which means you can "play" the LFOs chromatically from a connected MIDI keyboard.**
- **At -100 the LFO Rates tracks the incoming MIDI Notes "backwards" chromatically, which means the LFOs run slower the higher up on the keyboard you play, and vice versa.**
 The center MIDI note, i.e. where the LFO Rates "intersect" with the MIDI Note, is C3.

If Tempo Sync is enabled for the LFO(s) you will be able to do cool wobble effects by playing various notes on your MIDI keyboard, see "Tempo Sync".

Modulation inputs and outputs



LFO 1&2 input sections



Rate

Use this for dynamically modulating the Rate of the corresponding LFO. Attenuate the input signal with the knob.

Phase

Use this for dynamically modulating the Phase of the corresponding LFO. Attenuate the input signal with the knob.

Shuffle

Use this for dynamically modulating the Shuffle amount. Attenuate the input signal with the knob.

Amount

Use this for dynamically modulating the Level of the corresponding LFO. Attenuate the input signal with the knob.

LFO 1&2 output sections



CV

There are two CV signal outputs for each LFO, plus two additional outputs for the CV signal phase inverted.

Audio

There are two Audio signal outputs for each LFO, plus two additional outputs for the Audio signal phase inverted.

The difference between the Audio outputs and the CV outputs is that the signals on the Audio outputs have higher quality for use in e.g. audio processing applications.

Output LFO 1+2



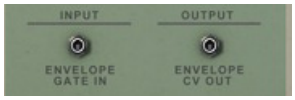
CV

There is one CV signal output for the LFO 1 and 2 signals summed with each other.

Audio

There is one Audio signal output for the LFO 1 and 2 signals summed with each other at audio quality.

Envelope connections



Envelope Gate In

A CV signal with a value > 0 present on this input will gate the Envelope (see "Envelope"). When the CV input signal decreases to zero or below, the gate is opened (deactivated).

Envelope CV Out

There is one CV signal output for the Envelope signal.

Tips and Tricks

Patch between LFO 1 and LFO 2 on the back for more flexibility

- **By connecting one LFO CV output to the Phase CV input of the other LFO you can achieve some very interesting rhythms. Try using a square wave and set the CV Trim rotary to around 25%.**
- **Try connecting an LFO CV output to the Shuffle CV input of the other LFO. Use the sawtooth waveform, at a slow Rate, in the first LFO to gradually add more shuffle over time.**
- **With the Envelope CV Out you can control both Shuffle and Phase from the Envelope and with the Envelope CV In you can trigger this effect with any CV signal.**

Using Pulsar as a monophonic synth

Because the LFO 1&2 can work in audio rate and can be tracked from a MIDI keyboard, it's possible to use Pulsar as a two oscillator mono synth with a lot of character. Here's how to set this up:

1. **Create Reason Rack Plugin as an instrument and add a Pulsar device.**
2. **Make sure the level of the LFO(s) you want to use is completely turned down.**
3. **Connect the Audio output of the LFO(s) you want to use to the I/O device.**
4. **Turn KBD Follow up to 100%.**
5. **Turn up the Level knob(s) in the Envelope section for the LFO(s) you want to use.**
 This decides how much the level of the LFO(s) will be affected by the envelope when you play.
6. **You can now play Pulsar from your MIDI keyboard!**

! **You have to tune Pulsar manually by adjusting the Rate(s) of the LFO(s) you are using. You can set the Rate knob(s) to max (1.05kHz) to be completely in tune - however, the Pulsar will then be 2 octaves above played MIDI note. Lower octaves can be found at around 524Hz and 263Hz (hold [Shift] to fine tune).**

! **Note that Pulsar is an LFO device and therefore doesn't feature the standard sound improvement features of a synth device (anti-aliasing filters etc.).**

► **By using LFO 2 to LFO 1 Rate modulation you can modulate the frequency of LFO 1, resulting in frequency modulation (FM).**

► **By using LFO 2 to LFO 1 Level modulation you can modulate the level of LFO 1, resulting in amplitude modulation (AM).**

► **By modulating the Shuffle you can achieve a sound similar to pulse width modulation (PWM), great for harsher tones!**

Chapter 48
RPG-8 Arpeggiator

Introduction



An arpeggiator generates rhythmic note patterns (arpeggios) from notes or chords. The RPG-8 Arpeggiator doesn't generate sound on its own, but has to be connected to another instrument device (just like the Matrix). It works by converting MIDI note data (input to the RPG-8) to Note CV (pitch) and Gate CV (note on/off plus velocity) signals. These CV/Gate signals are sent to the corresponding Sequencer Control inputs of an instrument device.

In addition to standard arpeggiator features the RPG-8 is equipped with a 16 step pattern editor for creating rhythmic variations.

The RPG-8 is monophonic and can control one voice in a connected instrument device.

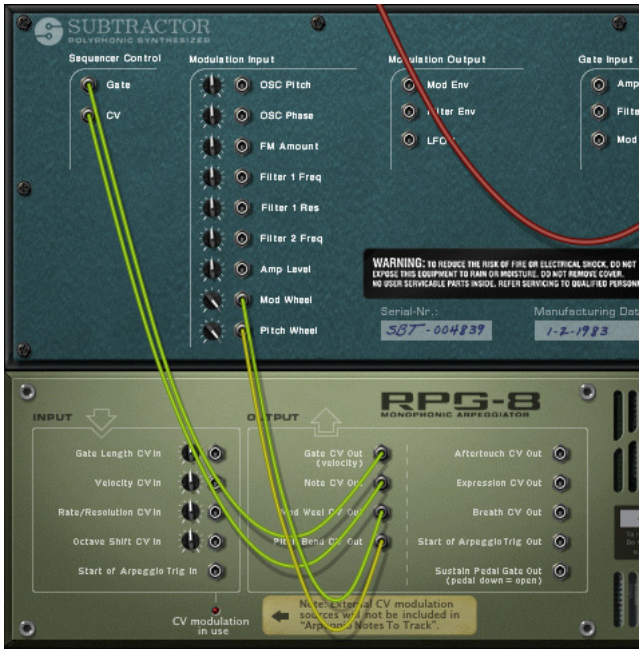
! **Please, note that this device is not available in Reason Lite Rack Plugin.**

Using the RPG-8

Setting up

The basic procedure is to input note data, either live or recorded, to the RPG-8 device. This note data is in turn sent to a target device via its Sequencer Control CV/Gate inputs. The resulting output from the target device can either be arpeggiated notes or simply mirror what is played on your control surface device. Proceed as follows:

1. **Create a Combinator device.**
2. **Drag an instrument device, e.g. a Subtractor to the Combinator "container".**
 Select a suitable patch, preferably one with a short attack time.
3. **Drag an RPG-8 Arpeggiator from the Utilities palette and drop below the Subtractor.**
 The RPG-8 Note and Gate CV outputs will be auto-routed to the instrument device Sequencer Control Gate and CV inputs, as you can see if you flip the rack around. In addition, the Mod Wheel and Pitch Bend CV outs are also auto-routed to the corresponding modulation inputs on the target device.



4. **Flip the rack and click the "Show Programmer" button on the Combinator panel.**

5. **Click the Subtractor in the Combinator display and deselect “Receive Notes”.**
 Since all instrument devices and such automatically gets MIDI input in Reason Rack Plug, you would get MIDI notes both to RPG-8 and Subtractor here. By deselecting “Receive Notes” for the Subtractor, only the RPG-8 will receive MIDI.



6. **Make sure the Arpeggiator Enable (“On”) button on the upper part of the panel is activated.**



7. **Play a few notes.**
 The notes in the chord you play are now arpeggiated for as long as you hold down the keys. The arpeggio will change directly if you release all notes and play another note or chord. If you add notes while holding down a chord, the arpeggio will continue with the added notes.



- **The display shows the notes played by the arpeggio pattern, with small bars indicating pitch for each step. The display is continuously updated as you play.**

- **The arpeggio will play in sync with song tempo by default in new RPG-8 devices.**
- **You do not need to start playback to generate arpeggios.**

→ **Try changing the arpeggiator mode using the Mode knob.**
The various modes govern how the notes are arpeggiated. For example "Up" means from lowest note to highest note, Up+Down from lowest to highest note, then back down to lowest note again. For a description of all the modes see "Mode switch".

→ **By activating the Hold button the arpeggio will continue to play even if you release the keys.**
If you play a new chord the arpeggio will continue to play, using the new notes. To stop the arpeggio, deactivate the Hold button or hit Stop on the transport. See "Hold On/Off".

→ **The "Insert" buttons can be used to introduce further variations to the arpeggio.**
See "Insert buttons".

→ **If you change the Rate parameter the rate of the arpeggiated notes will follow the song tempo at the selected note value resolution.**
Straight, triplet and dotted note values are available. The arpeggiator can also be free running, i.e. not synced to tempo - see "Rate".

→ **The Gate Length knob allows you to adjust the length of the arpeggiated notes.**
If the knob is turned fully clockwise ("Tie"), the gate is always open (the notes will be played legato). If it is turned fully counter-clockwise, the gate is closed (no sound). See "Gate Length".

→ **By using the Octave buttons beside the Mode knob you can increase or decrease the octave range of the arpeggio.**
With "1 Oct" selected the arpeggiated notes will be those that you press down on the keyboard. If "2 Oct" is selected the range will be expanded so that the arpeggiated notes play over two octaves, and so on - see "Octave range buttons".

→ **You can transpose the arpeggio up or down in octave steps using the Octave Shift buttons.**
See "Octave Shift".

→ **By using the Velocity knob you can set the arpeggio to play with a fixed velocity value (selectable between 1 to 127) for all notes. Turning the knob fully clockwise to the "Manual" position means that the arpeggio notes will use the same velocity as the MIDI notes you input.**
By using the Manual mode and varying the velocity for individual notes the arpeggio will become more animated and rhythmic. See "Velocity".

That covers the basic principles of how to set up and use the RPG-8 to arpeggiate notes that you play in real time. But to make full use of the RPG-8 there are many further applications:

→ **You can introduce rests for more complex rhythmical arpeggios by using the Pattern editor.**
See "Pattern editor" for a description.

→ **You can use the RPG-8 as a MIDI to CV converter which allows you to freely assign common performance MIDI controllers like Mod Wheel and Aftertouch to control parameters - see "Using the RPG-8 as a MIDI to CV converter".**

RPG-8 Parameters

MIDI-CV Converter parameters



The MIDI-CV Converter section to the left contains parameters that affect the CV output from the RPG-8, regardless of whether the Arpeggiator section is activated or not. The following parameters are available:

Velocity

The Velocity knob can be used to set a fixed velocity value for the notes that are output via the Gate CV Out jacks on the back of the RPG-8. If you set the Velocity knob to a value between "0" and "127", the Gate CV Out will be fixed (at the set value) regardless of the velocity of the incoming MIDI notes.

Turning the knob fully to the right activates Manual ("Man.") mode (a LED is lit when activated). In Manual mode the velocity levels will be sent out via the Gate CV Out with the same velocity value as they are input, i.e. "what goes in, will come out". Manual mode is on by default in new devices.

There is also a "Velocity CV" input at the back. If this is connected to a controller source (a LFO modulation output for example), the output will be a merge between the Velocity setting and the applied CV modulation by the LFO - see "CV Inputs".

Hold On/Off

If the Hold parameter is activated (lit button), an arpeggio will continue to run even if you release all keys. It will continue to arpeggiate the last notes played until a new note-on is received.

- **If you continue to hold down at least one key when Hold is on, any new notes will be added to the existing arpeggio as opposed to starting a new arpeggio.**
- **If the Arpeggiator section is off, and the Hold function is activated, there will be no note-offs for incoming notes played (i.e. the CV Gate stays open).**
- **The Hold On/Off status responds to Sustain Pedal messages - as long as the pedal is pressed down, Hold will stay activated.**

Octave Shift

This allows you to transpose the RPG-8 Note CV output in octave steps. You can octave shift up or down 3 octaves.

Octave Shift can also be CV controlled.

Arpeggiator parameters



The middle section contains the Arpeggiator parameters that govern how the arpeggio is played. The following parameters are available:

Arpeggiator “On” button

This switches the Arpeggiator on or off.

Mode switch

This determines the direction of the arpeggio notes.

Mode	Description
Up	This will generate an arpeggio that plays from the lowest note to the highest note.
Up+Down	Notes are played from lowest note to highest, then from highest back down to the lowest. The very lowest and the highest arpeggio notes are not repeated. I.e. the arpeggiator will play the lowest note to the second highest note, then the highest note to the second lowest note.
Down	Notes are played from the highest note to the lowest note.
Random	The notes you input will be arpeggiated randomly.
Manual	Notes are arpeggiated in the same order they are played when input.

Octave range buttons

The Octave buttons allow you to set the octave range of the arpeggio.

Use as follows:

Octave range	Description
1 Oct	The arpeggiated notes will be those that you press down on the keyboard.
2 Oct	If you select this, the arpeggio will be extended to a two octave range, i.e, the arpeggio pattern is played then repeated one octave up. In Random mode, the notes you input are played in random order over 2 octaves.
3 Oct	Same as 2 Oct but extended to a three octave range.
4 Oct	Same as 2 Oct but extended to a four octave range.

Insert buttons

Insert can be used to add variations to the arpeggio by repeating certain notes in a predetermined order. It works as follows:

Insert mode	Description
Off	No Insert repeat.
Low	The lowest note is repeated in between every second note in the arpeggio.
Hi	The highest note is repeated in between every second note in the arpeggio.
3-1	The arpeggio will play 3 notes forward, then step 1 note back and from there play 3 notes forward etc.
4-2	The arpeggio will play 4 notes forward, then step 2 notes back and from there play 4 notes forward etc.

Rate

This sets the rate of the arpeggio. There are two basic modes for the Rate parameter:

- **If Sync is activated, the Arpeggiator will play in sync with the sequencer tempo. By changing the Rate you can make the Arpeggiator play in different tempo resolutions in relation to the tempo setting.**
 Straight, dotted or triplet note values are available in 1/2 to 1/16 resolutions. In addition, there are also 1/32, 1/64 and 1/128 (straight) note resolutions.
- **If the "Free" button is activated, the arpeggio rate is free running, and not synced to tempo.**
 The Rate is then selectable from 0.1 to 250Hz.

Gate Length

This determines the length of the arpeggio notes. Minimum value is 0 (Gate closed - no output). Maximum value is "Tie", meaning the gate is open all the time. This parameter can be controlled via CV.

Single Note Repeat

Single Note repeat governs how the arpeggiator behaves when the user plays single keys or monophonic lines.

- **When Single Note Repeat is on, a single key will retrigger the gate, meaning the note will repeat.**
 If the Octave setting is 1 Oct, the note will simply repeat (given the Gate Length setting is not set to "Tie"). If the Octave setting is set to anything else, the note will repeat according to the Octave, Mode and Insert settings.
- **When this is off, single notes will not repeat and RPG-8 will play arpeggios when the user plays more than one key (chords).**

Shuffle

Shuffle is a rhythmic feature, that gives the arpeggio a more or less pronounced swing feel. It works by delaying all sixteenth notes that fall in between the eighth notes.

In the RPG-8 you can switch Shuffle on or off using the corresponding button. However, the amount of shuffle can be set globally (for all devices that incorporate this feature) with the Shuffle control in the I/O device.

Pattern editor

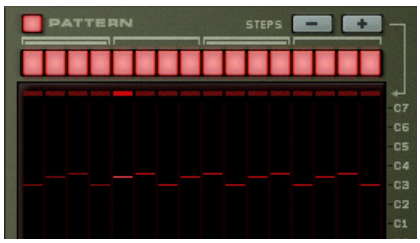


The Pattern editor allows you to introduce rests for arpeggio steps which can produce more rhythmic results. The Pattern editor has 16 step buttons at the top, and a main grid display where the arpeggio notes are represented as horizontal bars for each step in the arpeggio. The pitch of the arpeggio notes are shown on the vertical axis. Notes within the C-1 to C7 octave range are shown. Notes cannot be edited in the display, they are only a visual representation of the arpeggio.

- **The Pattern editor is activated with the "Pattern" button.**
 When activated, the Pattern button and the 16 Step buttons light up.
- **When you play a chord (or in case you have recorded notes, when you start playback) the arpeggio will play according to the current Arpeggiator parameter settings, as normal.**
 The only difference is that a pattern will be repeated in the display so that all 16 steps play the pattern.



A three note chord with Pattern off...



...and with Pattern on.

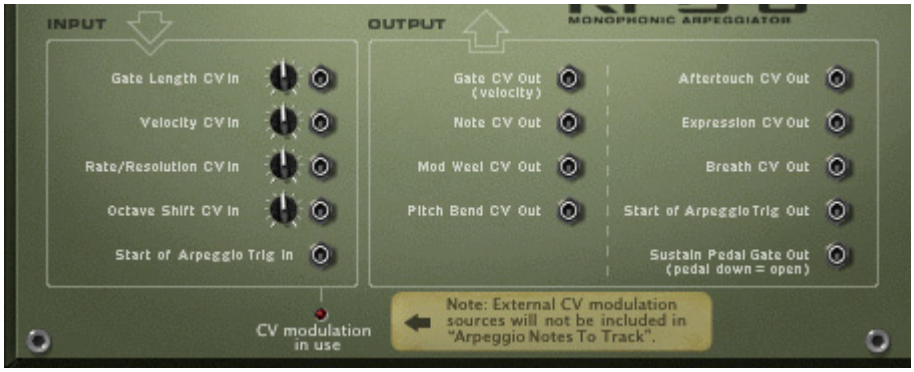
- **If you click on a step button it goes dark. This means that this step will insert a rest in the arpeggio pattern.**
 Note that no arpeggio notes are "skipped". Inserting a rest means that the step will be silent and the next active step will play the next note in the arpeggio pattern.
- **The "Steps -/+" buttons can be used to set the number of steps in the Pattern editor.**
 E.g. if you press the "Steps minus" button four times the last four step buttons will go dark and the Pattern editor cycle will start over after step 12.

Pattern functions

When the Pattern editor is activated, you will find some specific pattern functions on the Edit menu (and on the device context menu). These are as follows:

Function	Description
Alter Pattern	The Alter Pattern function modifies existing step patterns. Note that there has to be a pattern to start with - using the Alter function if all step buttons are active (or inactive) won't do anything.
Randomize Pattern	The Randomize Pattern function creates random patterns.
Invert Pattern	This will invert the pattern, i.e. active steps will become rests and vice versa.
Shift Pattern L/R	The Shift Pattern functions move the pattern one step to the left or right.

CV connections



On the back of the RPG-8 you can find a number of useful CV connectors. These are as follows:

CV Inputs

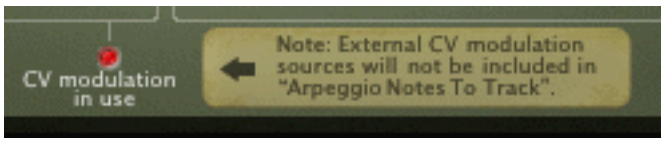
There are five CV inputs, of which four can be used to control RPG-8 parameters that have associated controls on the front panel. These parameters are Gate Length, Velocity, Rate and Octave Shift.

If you use an external source to modulate these parameters, the incoming CV is merged with the setting on the front of the device.

An example: Velocity is set to 50 on the front panel. A Matrix (bi-polar curve) that varies between +- 20, with the voltage trim pot set to 64 (50%) is connected to the Velocity CV input. The resulting Velocity should then vary between 40-60.

In addition to the above CV inputs, there is a "Start of Arpeggio Trig In" connector. This restarts the arpeggio figure from step 1 when this input receives a gate trigger. See "Triggering arpeggios" for a tip on how this can be used. If something is connected to this input the RPG-8 will not generate arpeggios unless a Gate trigger is received.

Note: If you are modulating the arpeggio using the CV Input jacks, this will not affect the rendered arpeggio notes.



CV Outputs

The following CV outputs are available:

- **Gate CV Out (Velocity) transmits a gate/velocity value that corresponds to the Velocity parameter setting.**
 This is normally auto-routed to the Sequencer Control Gate input on an instrument device.
- **Note CV Out transmits the notes generated by the Arpeggiator (or if this is off, the notes you input to the RPG-8).**
 This is normally auto-routed to the Sequencer Control CV input on an instrument device.
- **Mod Wheel/Pitch Bend Outputs will normally connect to the corresponding inputs of the instrument device controlled by RPG-8.**
- **The common MIDI performance controllers Aftertouch/Expression/Breath can be used to control any parameters using these CV outputs.**
- **Start of Arpeggio Trig Out - every time the arpeggio figures start over this output will send a gate signal.**
 This can be used to trigger filter, amp and mod envelopes in other devices, e.g. Subtractor. The length of this gate is the same as the length of the first note in the arpeggio.
- **Sustain Pedal Gate Out - the sustain pedal can be used to trigger envelopes in other devices.**
 If you plug a cable into the "Sustain Pedal Gate Out" the normal pedal to the Hold parameter connection is broken and the sustain pedal function is changed to send out a gate signal with velocity equalling the Velocity setting knob (if Manual mode is on then velocity defaults to a value of 100).

Tips and tricks

Using the RPG-8 for modulation

You can use the RPG-8 as a modulation source, much like an LFO. The RPG-8 can generate stepped modulation that is both synchronized to tempo and controlled by note input. In this and in following examples we assume you already have a RPG-8 connected to an instrument device.

- **With the RPG-8 selected in the rack, create a Spider CV Merger/Splitter.**
 The RPG-8 Note and Gate CV outputs are auto-routed to the Spider Split A and B inputs, and the Spider first split Note and Gate outputs are connected to the instrument device (as they were before creating the Spider).

You now have 2 (and 1 Inverted) additional Note and Gate CV outputs via the Spider.

- **Try using the Note CV split outputs to modulate other parameters in the instrument device, e.g. filter frequency.**
 The filter frequency will then track the notes generated by the Arpeggiator.
- **You can of course also connect the Note CV out to various parameters in any device, not just the device connected to the RPG-8.**
- **You can connect the Spider split CV/Gate outputs to the corresponding Sequencer Control CV inputs to another instrument device so that the RPG-8 controls two (or more) devices. (This can of course also be done by connecting the RPG-8 to a Combinator device.)**

Triggering arpeggios

On the back panel there is a "Start of Arpeggio Trig In" CV connector. This restarts the arpeggio figure from step 1 when this input receives a gate trigger. You could use this in the following way:

- **One or more Redrum channels Gate out can reset the step pattern to create rhythmic patterns in sync.**
- **You could use the Matrix in the same way - each positive Gate signal will restart the arpeggio figure.**

! **Note that no arpeggio will be generated unless a Gate trigger is received when something is plugged in to the "Start of Arpeggio Trig In" CV connector.**

Triggering samples

The Gate CV output can be used to trigger samples, either in Redrum or Kong or in the NN-19 or NN-XT Sampler.

- **Connect the RPG-8 Gate CV out to the Gate (Sequencer Control) in on the NN-19/NN-XT or to one of the individual Gate Channel inputs of Kong or Redrum.**
 Gate values will now trigger the sample on each step with Gate values above "0".

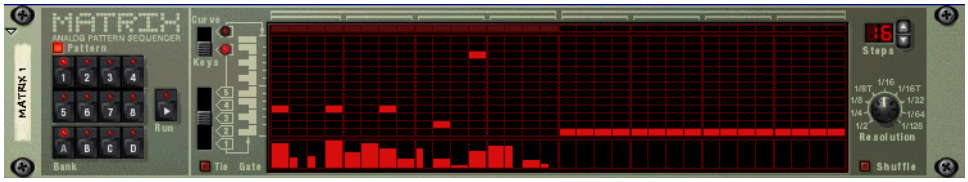
Using the RPG-8 as a MIDI to CV converter

You can also use the RPG-8 as a stand-alone MIDI to CV converter without generating arpeggios. In this mode (Arpeggiator Off) you can play instrument devices just like as if MIDI input was directly connected to the target device.

The main benefit of this mode is that you can use note pitch and velocity to control parameters, not only in the target device but in any device.

Chapter 49
Matrix Pattern Sequencer

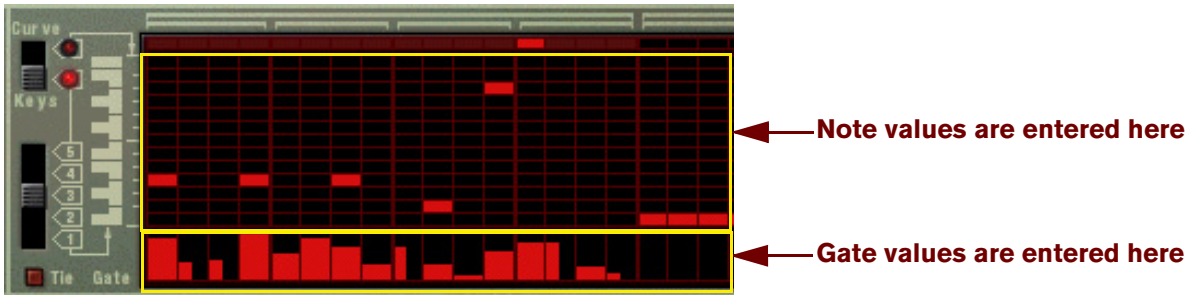
Introduction



The Matrix is a pattern-based device. Matrix doesn't generate sound on its own, but has to be connected to another instrument device. It basically works by sending pattern data in the form of Note CV (pitch) and Gate CV (note on/off plus velocity) or Curve CV (for general CV parameter control) signals to a device or device parameter. The patterns can be up to 32 steps, and there are 32 memory locations for storing pattern data. The Matrix is monophonic and can control one voice in an instrument device.

Unlike most other devices in Reason, the user interface of the Matrix is not modeled on any existing hardware equivalent. The hardware devices that could be said to have similar functionality are analog step sequencers, which usually had rows of knobs that controlled the note pitch and gate values for each step.

About the three Output types



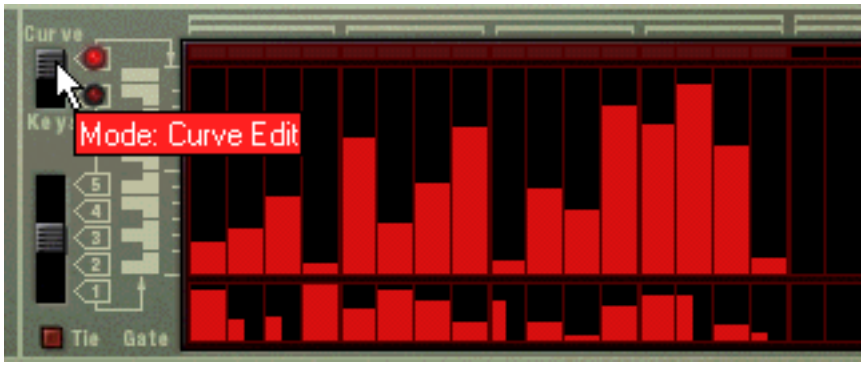
Note and Gate CV values.

The Matrix can produce three types of output: Curve CV, Note (Key) CV and Gate CV.

- **Note CV normally controls note pitch.**
 When connected to an instrument device Sequencer Control input, the values correspond to semitone steps.
- **Gate CV represents a note-on/off value, plus a *level* value (that could be likened to velocity).**

Both of these two outputs are typically connected to the Sequencer Control Gate and CV inputs on a compatible instrument device. For example, if you create a Matrix with either a synthesizer (Subtractor, Malström) or a sampler (NN-19, NN-XT) selected, they will be auto-routed in this way, and will control one voice in the device.

- **Curve CV is a separate pattern, programmed separately from the Note/Key and Gate CV.**



Curve CV values (upper window).

This is useful for programming CV curves that control other parameters other than note pitch (although you could do this too). This way you could control the note pitch and triggering from the Key and Gate outputs for a device, then add a second independent pattern using the Curve CV output that could control filter cutoff for example.

It should be stressed that all three outputs can be used in any number of ways. For example, you could use the Gate CV to trigger a drum in Redrum, or let the Curve CV control the feedback parameter of a delay, etc.

Programming patterns

Pattern basics

Matrix contains a built-in pattern sequencer that repeatedly plays back a pattern of a specified length. The typical example in the "real world" is a drum machine which plays drum patterns, usually one or two bars in length.

Having the same pattern repeat throughout a whole song may be fine in some cases, but most often you want some variations. The solution is to create several different patterns and program pattern changes (automatic switching from one pattern to another) at the desired positions in the song.

Selecting Patterns

The Matrix has 32 pattern memories, divided into four banks (A, B, C, D).



The Bank and Pattern buttons for the Matrix pattern sequencer.

→ **To select a pattern in the current bank, click on the desired Pattern button (1-8).**

→ **To select a pattern in another bank, first click the desired Bank button (A, B, C, D) and then click the Pattern button.**
Nothing happens until you click the Pattern button.

The pattern change takes effect on the next downbeat according to the time signature set in the transport panel or on the transport track.

The Pattern Enable switch



The Pattern Enable switch.

Next to the Bank and Pattern buttons you will find an additional switch, which is normally activated. If you click this to turn it off, the pattern playback will be disabled, starting at the next downbeat - exactly as if you had selected an empty (silent) pattern. For example, this can be used for bringing Matrix patterns in and out of the mix during playback.

Steps

Matrix patterns consist of a number of discrete steps. For each step, you can enter a note, a CV value and a Curve value. When you run the pattern, each step will be played back in turn and will play a sound or send out the information programmed for this step. If you have ever used a drum machine, this will be obvious to you.

Clearing a Pattern

To clear (empty) a pattern, select it and use the Clear Pattern command on the Edit menu or device context menu.

! **Note that clearing a pattern doesn't affect the pattern length, resolution or shuffle settings!**

Using Cut, Copy and Paste

By using the Cut, Copy and Paste Pattern commands on the Edit menu or device context menu, you can move or duplicate patterns. The following rules apply:

→ **Copy Pattern makes a copy of the currently selected pattern and places the copy on the clipboard.**

→ **Cut Pattern moves the currently selected pattern to the clipboard.**
This is the same as first performing Copy Pattern and then Clear Pattern.

→ **Paste Pattern copies the pattern on the clipboard to the selected pattern location in the selected device.**
This overwrites the selected pattern with the one on the clipboard.

Transferring patterns between Reason Rack Plugin instances

If you want to copy patterns between different Reason Rack Plugin instances songs, you can use copy and paste:

1. **Select the pattern you want to copy in the first instance.**
2. **Select Copy Pattern from the device context menu.**
3. **In the other instance select the bank and pattern location to which you want to copy the pattern.**
Note that any pattern already stored in that location will be overwritten!
4. **Select Paste Pattern from device context menu.**

Tutorial

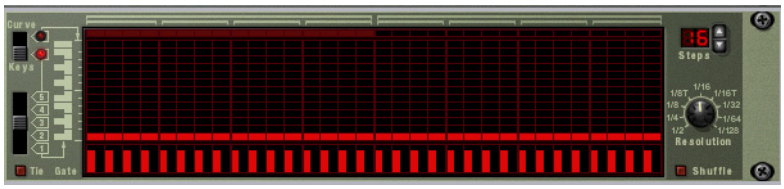
The programming procedure of the Matrix is to input note and gate values into the upper and lower fields of the pattern window respectively. You can input values by clicking or dragging in the pattern window. Proceed as follows:

1. **Create a Subtractor synth.**
 You don't have to use the Subtractor device to use the Matrix, in fact you don't have to use an instrument device at all, but for this basic tutorial we will use a "standard" setup.

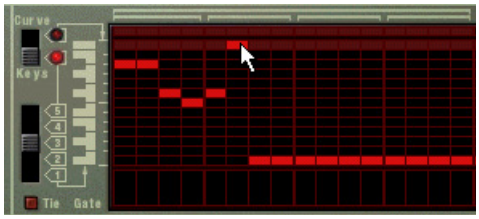
2. **With the Subtractor selected, create a Matrix Pattern Sequencer.**
 The Matrix Note and Gate CV outputs will now be auto-routed to Subtractors Sequencer Control Gate and CV inputs, as you can see if you flip the rack around.



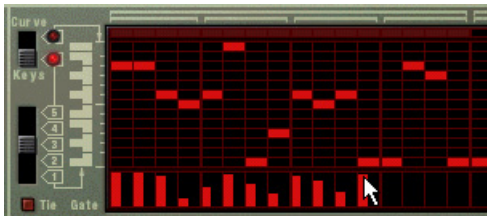
3. **Make sure that the switch to the left of the pattern window is set to "Keys" position.**
 As you can see, there are two rows of red rectangles. The one with horizontal rectangles at the bottom of the upper field in the pattern window represent note pitch, for each step in a pattern. At the moment they are all set to the same note pitch. The row of vertical rectangles in the lower field represent Gate velocity values - currently these are all set to a velocity value of 100 for all steps.



4. **Click inside the upper grid section of the Matrix pattern window.**
 An orientation line is displayed in the grid to make it easier for you to find the desired note, and the red rectangles are placed according to where you click. You can drag to input continuous note values.

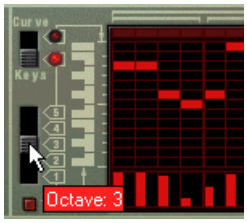


5. **Click and drag in the lower area of the pattern window.**
 You can create vertical Gate velocity strips of varying heights. The higher the strip, the higher the velocity value.



6. **Press the Play button on the Matrix.**
 The pattern you "programmed" in the previous steps is now repeated. At the top of the pattern window, a red rectangle indicates every step of the pattern.

- **If you now click or drag in the upper grid section with the pattern playing, you can hear how the note pitches change.**
 The note pitch corresponds to the keyboard printed to the left of the pattern window, in a one octave range, and as previously mentioned, an orientation line is visible when clicking or dragging, making it easy to find the note pitch on the keyboard.
- **If you now click or drag in the lower gate section while the pattern is playing, you can hear how the timbre and volume changes.**
- **If you drag some of the vertical rectangles down so that they disappear from view, the corresponding steps of the pattern are completely silenced.**
- **By using the 5-way switch below the "Keys/Curve" switch you can input notes in other octave ranges (over five octaves).**
 Note that there can only be one note for each step in the pattern.



7. **By using a combination of the methods described in the above steps, you can program suitable note values for each step, decide which steps should be played and set their velocity with the gate values.**

Using Curve Patterns

Curve patterns are independent patterns that can be applied separately to the note pattern programmed in "Keys" mode. If you switch the Keys/Curve switch to "Curve", the note, but not the gate steps, disappear from view, and leaves the upper area of the pattern window empty. You can now start programming a curve pattern. Proceed as follows:

1. **Draw a curve, using the same method as for notes or gates.**
 As you can see, the Curve pattern looks like large vertical gate steps.

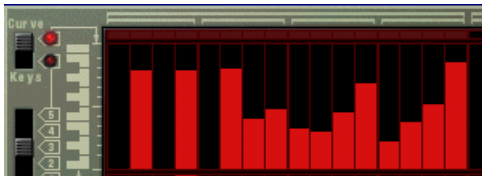
→ **If you play the pattern, nothing has changed, i.e. the pattern sounds exactly like it did before the Curve pattern was drawn.**
This is because the Curve CV output hasn't been connected to any parameter yet.

2. **Flip the rack around so you can see the back panel of the Matrix.**
3. **Connect the Curve CV output to the Filter Cutoff Modulation Input on the Subtractor.**
 Now the curve pattern controls the filter frequency of the Subtractor.

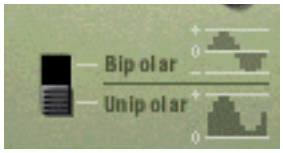
► **If the effect isn't very noticeable, try raising the filter Q parameter, and lowering the filter frequency.**

→ **The Curve CV output can be connected to any device CV or Modulation input.**
Actually, Curve CV signals can also produce Gate triggers (used for triggering samples or envelopes for example).

→ **A Gate trigger is produced for each curve pattern step that follows a value of "0".**
If you look at the picture below, steps 2,4 and 6 will produce a trigger, because steps 1, 3 and 5 are set to zero, but the rest of the pattern would not.

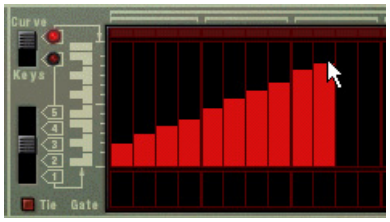


About Unipolar and Bipolar Curves



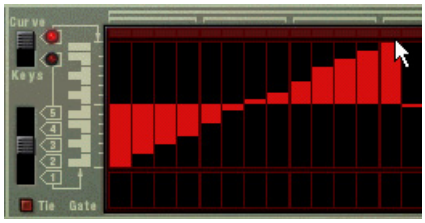
On the back panel of the Matrix you will find a switch, allowing you to select between "Unipolar" or "Bipolar" Curves. The difference is as follows:

→ **A unipolar curve has values starting from "0" and up.**
"0" is the value produced by all steps when they are "empty" (not visible). Unipolar is the default setting of this switch when a new Matrix is created.



Unipolar curve

- **A bipolar curve is divided in the "middle", with the middle representing a value of "0".**
 The curve reflects this. If no curve has been drawn and you switch to bipolar mode, all steps go from the bottom up to the middle of the scale printed to the left of the pattern window. Thus, all steps are at "0", and the curve can be drawn both up and down from the middle.



Bipolar curve

Bipolar curves are essential in some instances. If you want to use the Matrix to CV control the Pan parameter for a mixer channel for example, a unipolar curve would start at zero - which for Pan equals center position. This means that you would only be able to use the curve to pan in *one direction* from this center position. A bipolar curve however, will have the zero value in the middle, allowing you to draw pan curves in both directions. Bipolar curves can also be used for controlling parameters with positive and negative values.

Setting Pattern Length



You may want to make settings for Pattern length, i.e. the number of steps the pattern should play before repeating:

- **The "Steps" spin controls are used to set the number of steps you wish the pattern to play.**
 The range is 1 to 32. You can always extend the number of steps at a later stage, as this will merely add empty steps at the end of the original pattern. You could also make it shorter, but that would (obviously) mean that the steps you remove won't play back. The steps you remove aren't erased though, if you set the step number back again, anything recorded in the previously removed step locations will be played back.

Using Tied Notes

If you activate "Tie" to the left of the Gate pattern window, you can create longer notes (eighth notes, quarter notes etc.). A quick way to draw tied gates is to hold down [Shift] when you input the gate values.



Entering tied gate values.

- **Each step that has one tied gate value will be twice the length compared to a normal step.**
 Tied gate steps are indicated by being twice as wide in the pattern window.
- **If two or more notes of the same pitch are tied together, the result will be even longer notes.**

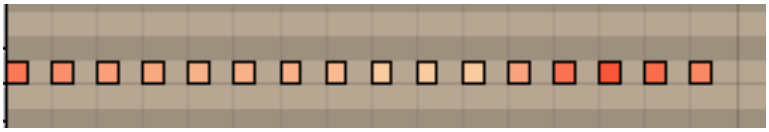
Tied notes are also essential if you want to create typical TB-303 "Acid"-type lead lines - see "Programming "Acid Style" lead lines".

Setting Pattern Resolution

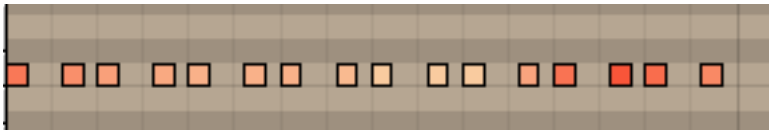
Matrix always follows the tempo setting on the transport panel, but you can also make Matrix play in different tempo "resolutions" in relation to the tempo setting.

Pattern Shuffle

Shuffle is a rhythmic feature, that gives the music a more or less pronounced swing feel. It works by delaying all sixteenth notes that fall in between the eighth notes.



Example of straight sixteenth note pattern (viewed in the main sequencer).



The same sixteenth note pattern with shuffle applied.

You can activate or deactivate shuffle individually for each pattern in a pattern device. However, the amount of shuffle is set globally with the Shuffle control in the I/O device.

Pattern Mute

If you deactivate the "Pattern" button above the Pattern select buttons, the pattern playback will be muted, starting at the next downbeat (exactly as if you had selected an empty (silent) pattern). For example, this can be used for bringing different pattern devices in and out of the mix during playback.

Pattern Functions

When a pattern device is selected, you will find some specific pattern functions on the Edit menu (and on the device context menu).

Shift Pattern Left/Right

The Shift Pattern functions move the notes and corresponding gate values in a pattern one step to the left or right.

Shift Pattern Up/Down

! This function does not alter the Curve CV. This is because the values produced by the Curve CV do not necessarily correspond to semitone note steps at all.

The Shift Pattern functions will transpose all the notes in a pattern one semitone up or down.

Randomize Pattern

The Randomize Pattern function create random patterns. These can often be great starting points and help you get new ideas. Both Note, Gate and Curve CV values will be created.

Alter Pattern

The Alter Pattern function modifies existing patterns. Note that there must be something in the pattern for the function to work on - using an Alter function on an empty pattern will not do anything.

! **Note that Randomize and Alter affects both the Gate, Note and Curve CV!**

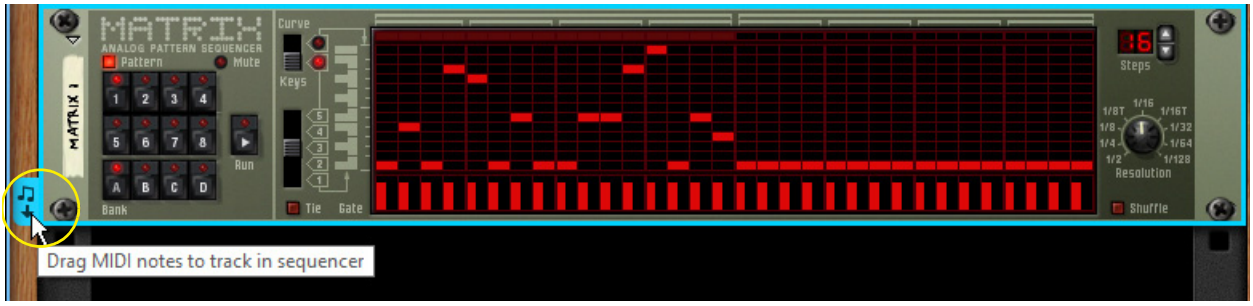
Chaining Patterns

When you have created several patterns that belong together, you will most probably want to make these play back in a certain order. This Pattern selection can be automated in the main sequencer, just like any other parameter automation.

Copy MIDI Notes to a sequencer track

It's possible to copy the MIDI Notes of the currently selected Pattern to a track in your DAW's sequencer:

1. **Click to select the desired Pattern.**
2. **Click the "Drag MIDI Notes to track in sequencer" icon and drag to the destination track in the sequencer:**



! **Be sure to disable the Enable Pattern Section function on the Matrix panel afterwards, to avoid "doubled notes" during playback.**

! **Curve patterns cannot be converted to sequencer data! Only the note pattern and the gate values will be converted.**

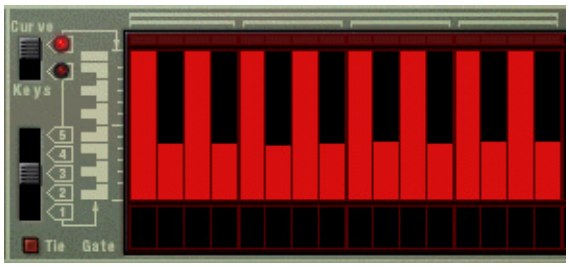
Example usage

As mentioned previously, the Matrix is a very flexible device. Here follows a few examples of how you can use the Matrix Pattern Sequencer.

Using the Matrix for modulation

You can effectively use the Matrix as a modulation source, much like an LFO. Just like the LFOs in Reason's instrument devices, the Matrix can generate modulation that is synchronized to tempo, which has many advantages. Proceed as follows:

1. **Create a Synthesizer (e.g. a Subtractor or Malström).**
2. **Create a Matrix Pattern Sequencer, or if one already exists, set it to an empty pattern.**
3. **Flip the rack around and disconnect the Gate and Note CV outputs.**
 Gate/Note CV is not used in this example
4. **Flip the rack around and connect the Curve CV output on the Matrix back panel to the "Amp Level" Modulation input on the synthesizer.**
 This parameter is used for modulating the output level (volume) of the synthesizer. Volume modulation is often referred to as Tremolo. You can use a unipolar curve (see "About Unipolar and Bipolar Curves") for this example.
5. **Flip the rack back again, and switch the Matrix to display the Curve pattern window.**
6. **Draw a curve like the one shown in the illustration below.**
 If you use fewer or more steps than16 (as shown in the picture), just draw the curve so that it roughly matches the shape in the picture.



7. **Activate Click on the Transport Panel.**
8. **Select the track that is routed to the synthesizer, so that you can play it from your MIDI keyboard.**
9. **Activate Play on the transport panel, and hold a chord down on your keyboard.**
 You should now hear the volume being modulated by the Curve pattern.
10. **While still in play mode, you can use the Resolution knob to change the modulation "rate" in relation to the tempo.**
 For each clockwise resolution step the modulation speed is doubled and vice versa, but it will always stay in sync with the tempo.

Programming “Acid Style” lead lines

By “acid style” lead lines we mean patterns that use a combination of Legato and slide (or portamento) effects to produce the widely used hypnotic “wavy” sound produced by the original Roland TB-303, and recreated in the Propellerhead Software product ReBirth. To approximate this typical sound using Reason, proceed as follows:

1. **Create a Synthesizer (Subtractor or Malström).**
2. **Create a Matrix Pattern Sequencer, or if one already exists, set it to an empty pattern.**
3. **Make sure that the Note and Gate CV outputs are connected to the synthesizers Sequencer Control CV and Gate inputs, respectively.**
4. **For Subtractor, select either a Init Patch, or use the “TB Synth” patch in the Monosynth category of the Reason Factory Sound Bank.**

→ **If you use an Init patch, it is important that you make the following settings:**

- **Set Polyphony to “1”.**
- **Switch Trigger Mode to “Legato”.**
- **Set Portamento to a value around “50”.**

5. **Create a pattern in Matrix, and keep it playing back.**

→ **If “Tie” (see “Using Tied Notes”) now is activated for a step, the note will be tied to the next and the pitch will continuously “glide” to the pitch of the following step.**
Please note that Tie should be activated on the note you wish to slide from, and not the note you slide to.

→ **If you have several tied notes, one after the other, they will play as one long legato phrase. This can be used to create “wavy” lead lines with pitch bend effects.**

6. **Experiment with different Note, Tie and Gate values.**
If you have ever used a TB-303 or ReBirth, you should now begin to get the hang of how you can create patterns in that particular style by using the Matrix together with a synthesizer.

← **Adding a DDL-1 (delay), and a D-11 (distortion) effect device will make it sound even more “ReBirth”-like, but of course you are also able to get a much wider range of timbres by utilizing Reason’s other sound and modulation capabilities.**

Triggering samples

The Gate CV output can be used to trigger samples, either in Redrum or in the NN-19 or NN-XT Sampler.

→ **Connect the Matrix Gate CV out to the Gate (Sequencer Control) in on the NN-19/NN-XT or to one of the individual Gate Channel inputs of Redrum.**
Gate values will now trigger the sample on each step with Gate values above “0”.

Chapter 50
Mixer 14:2

Introduction



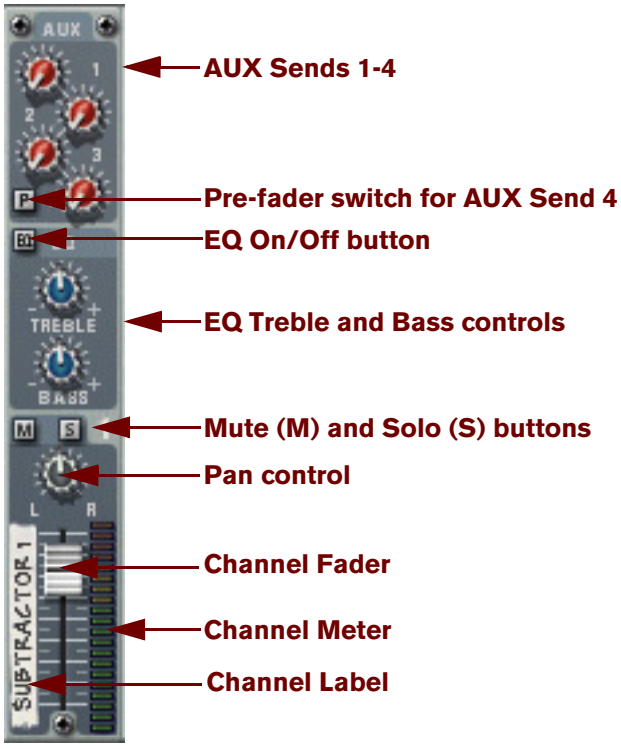
The Mixer 14:2 allows you to control the level, stereo placement (Pan), tone (EQ) and effect mix (AUX Sends) of each connected audio device.

If you have ever used a conventional hardware audio mixer, you will most likely find the Mixer very straightforward to use. It is configured with 14 (stereo) input channels, which are combined and routed to the Left and Right Master outputs. The vertical channel "strips" are identical and contain - from the top down - four Auxiliary Sends, an EQ section, Mute and Solo buttons, Pan control, and a Level fader.

Every mixer parameter can of course be automated.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

The Channel Strip



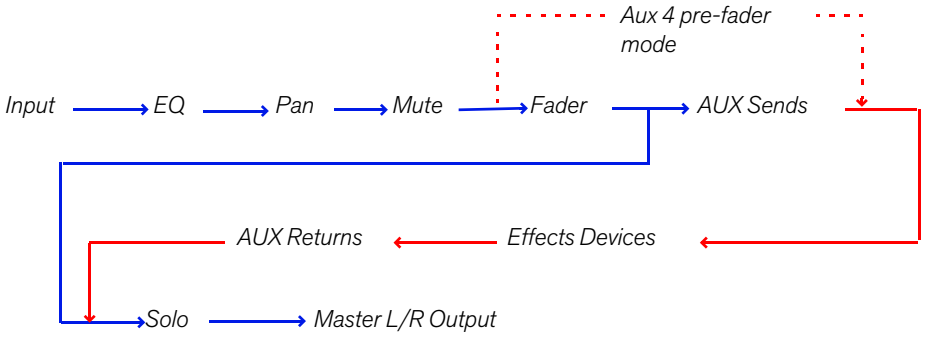
Each channel strip in the Mixer 14:2 contains the items listed in "Channel Strip Controls".

Channel Strip Controls

Item	Description	Range
Channel Fader	The channel fader is used to control the output level of each corresponding channel. By adjusting the faders, you can set the desired mix (balance) between different devices connected to the Mixer.	0 - 127
Channel Label	Each channel in the mixer that has a device connected to it, displays a read-only label with the name of the device to the left of the fader.	N/A
Channel Meter	The meter is a graphical representation of the channel output level. If the signal level pushes the meter into the range of the red area, try lowering either the output level of the device connected to the channel, or the channel fader itself, to avoid distortion.	N/A
Pan Control	Use this control to set the left/right position of the channel in the stereo field. [Command]/[Ctrl]-click the Pan knob to set Pan to the default "0" (center position).	-64 – 0 – 63
Mute (M) and Solo (S) Buttons	Clicking a channel's Mute button silences the output of that channel. Click the button again to unmute the channel. Clicking a channel's Solo button silences all other mixer channels, so that you only hear the soloed channel. Several channels can be soloed at the same time, but if this is the case, note that soloed channels can't be muted with the Mute button. To mute one of several channels in solo mode you simply "unsolo" it.	On/Off
EQ Treble and Bass controls	The EQ Treble and Bass controls is used to cut or boost the higher and lower frequencies of the signal, respectively. Click on the EQ button to activate the EQ. If you need more advanced EQ, you can always use a PEQ2 parametric EQ as an insert effect for a device. Note also the two EQ modes - see "About the EQ modes".	Treble: +/- 24 dB at 12 kHz. Bass: +/- 24 dB at 80 Hz.
Auxiliary (AUX) Effect Send 1-4	The four independent AUX Sends control the amount of channel signal that is to be sent to other devices - typically effect processors. The effect output is then normally returned to the Mixer via the AUX Return inputs (see "The Auxiliary Return Section") where it is mixed with the dry (non-processed) signal. If you create an effect device when the Mixer is selected, the effect is auto-routed to the first available Send/Return connectors. You can then control the amount of effect that is to be applied to any device connected to a Mixer channel via the corresponding AUX Send knob. The AUX Send outputs are taken post channel fader, but you have the option of selecting Pre-fader mode for AUX Send 4 (by clicking the "P" button next to the send so that it lights up). In that mode, the send level is independent of the channel fader. The sends are in stereo but can be used in mono as well.	0 - 127

The Mixer signal flow

The basic signal flow for a channel in the Mixer 14:2 is as follows:



Note that the Solo function is true “in-place” solo, meaning that if the channel uses Auxiliary sends routed to effect devices, the soloed output signal will also include the soloed channel(s) including any Aux Send effects.

Note also that if the pre-fader send mode is activated for Aux 4 the send is tapped after the EQ and Pan controls but before the channel fader.

About the EQ modes



With Reason 2.5, the EQ modules in the Mixer were improved to get an even better sound and character. However, if you want to emulate the character of older Reason versions, you may want to use the “old” EQ mode to ensure that it sounds exactly the same.

On the back of the Mixer 14:2 you will find a switch for this - select “Improved EQ” for the new EQ types or “Compatible EQ” for the old-style EQ. The parameters are exactly the same in both cases.

The Auxiliary Return Section



The Auxiliary Returns provide an “extra” four stereo inputs in addition to the Mixer 14:2’s 14 stereo channels. The main function of Return channels is to provide inputs for connected Send effects devices. Each Aux Return channel has a level control, and a read-only tape label that display the name of the device connected to the Return channel.

The Master Fader



The Master L/R fader controls the summed output level of all channels in the Mixer 14:2. Use this to change the relative level of all channels, to make fade-outs etc.

Connections

All input and output connectors are as usual located on the back panel of the Mixer 14:2. Special connectors are used for “chaining” two or more Mixer 14:2 devices together. This is described on “Chaining several Mixer 14:2 devices”.

Mixer Channel Connections

- **Each mixer channel features stereo left/right inputs for connecting audio devices.**
 Use the left input when manually connecting a mono signal source.
- **In addition, there are two Control Voltage (CV) inputs (with associated voltage trim pots), for voltage controlling channel Level and Pan from other devices.**

Auxiliary (AUX) Send Out



→ **There are four stereo Send Out connectors, which normally are used to connect to the inputs of effect devices.**
To connect a send to a mono-input device, use the Left (Mono) output.

When a Send is connected to an effects device, the corresponding AUX Send knob determines the level of the signal sent to the effect device for each channel. The Send Output is taken post-channel fader but you have the option of selecting pre-fader mode for AUX Send 4.

→ **Note that some effects (for example the Comp-01 compressor or the PEQ2 parametric EQ) are effect types which are not designed to be used as AUX Send effects, but rather as insert effects, where the whole signal is passed through the effect.**
Alternatively, you could use AUX Send 4 in pre-fader mode and lower the channel fader completely.

Auxiliary (AUX) Returns



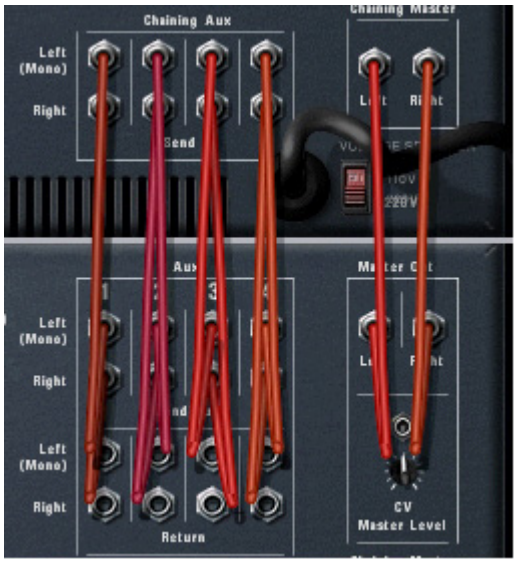
→ **There are four stereo Return input connectors.**
These are normally connected to the left and right outputs of effect devices.

Master Left/Right Outputs



- **The Master outputs are auto-routed to the first available output pair in the I/O device.**
If used in a Combinator, the Master Outs are normally connected to the “From Devices” connectors on the Combinator.
- **In addition to the Master Out connectors, there is a Control Voltage (CV) input (and an associated trim pot), for voltage controlling the Master Level from another device.**

Chaining several Mixer 14:2 devices



Two chained Mixer 14:2 devices are connected like this, the top Mixer being the "Master" Mixer.

If you want more Mixer channels, you can chain several Mixer 14:2 devices.

→ **Select the existing Mixer 14:2 device and choose "Create:Mixer 14:2" from the context menu.**
The new Mixer is automatically connected via the "Chaining Master" and "Chaining Aux" connectors of the selected Mixer.

- **The newly created Mixer's Master Output is connected to the original Mixer's Chaining Master input.**
The Master Out Level for the new Mixer is now controllable from the original Mixer's Master fader - so that this fader now controls the Master output level of both mixers.
- **The newly created Mixer's four stereo Aux Send outputs is connected to the original Mixer's Chaining Aux connectors.**
The new Mixer will now have access to any Aux Send effects connected to the original Mixer, via the same corresponding Aux Send(s).

This way, the two Mixers operate as "one".

! **One exception is the Mute/Solo function, which is not chained. Thus, soloing a channel in one of the Mixers, will not mute the channels in the other Mixer.**

You can create as many Mixers as you like, they will be chained in the same way, with one Mixer remaining the "master" (i.e. it controls the Master level of all chained Mixers and supplies the Aux Send effect sources).

Partially or Non-Chained Mixers

You can also have several Mixers that are only partially or not chained at all.

- **You may for example wish to have different Aux Send effects for one Mixer.**
Then simply disconnect one or more of the Send Out to Chaining Aux connectors, and assign new Send effects.

→ **You could for example send the Master output of one Mixer to another output pair on the I/O device, instead of the Chaining Master inputs.**

Chapter 51
The Line Mixer 6:2

Introduction



The Line Mixer 6:2 allows you to control the level, stereo placement (Pan) and effect mix (AUX Send) of each connected audio device.

The Line Mixer is configured with 6 (stereo) input channels, which are combined and routed to the Left and Right Master outputs.

Channel parameters

The channels are identical and contain an Auxiliary Send, Mute and Solo buttons, a Pan control, and a Level control:

Item	Description
Level control	This controls the output level of each corresponding channel, allowing you to set the desired mix (balance) between different devices connected to the Line Mixer.
Channel label	Each channel in the mixer that has a device connected to it, displays a read-only label with the name of the device.
Channel meter	The meter is a graphical representation of the channel output level. If the signal level pushes the meter into the range of the red area, try lowering either the output level of the device connected to the channel, or the Level control itself, to avoid distortion.
Pan control	Use this control to set the left/right position of the channel in the stereo field. [Ctrl]-click (Win) or the Pan knob to set Pan to the default "0" (center position).
Mute (M) and Solo (S) Buttons	Clicking a channel's Mute button silences the output of that channel. Click the button again to unmute the channel. Clicking a channel's Solo button silences all other mixer channels, so that you only hear the soloed channel. Several channels can be soloed at the same time. If this is the case, note that soloed channels can't be muted with the Mute button. To mute one of several channels in solo mode you simply "unsolo" it.
Auxiliary (AUX) Effect Send	The AUX Send controls the amount of channel signal that is to be sent to other devices - typically effect processors. The effect output is then normally returned to the Mixer via the AUX Return input where it is mixed with the dry (non-processed) signal. If you create an effect device when the Mixer is selected, the effect is auto-routed to the Send/Return connectors. You can then control the amount of effect that is to be applied to any device connected to a Line Mixer channel via the AUX Send knob. The Send can be taken pre or post channel level - see "Auxiliary (AUX) Send".

The Auxiliary Return section

The AUX Return channels provide inputs for a connected send effect device. The Aux Return channels have a single level control on the front panel.

Master level

The Master L/R fader controls the summed output level of all channels in the Mixer. Use this to change the relative level of all channels, to make fade-outs etc.

Connections

All input and output connectors are as usual located on the back panel of the Line Mixer.

Mixer Channel Connections

- **Each mixer channel features stereo left/right inputs for connecting audio devices.**
 Use the left input when manually connecting a mono signal source.
- **In addition, there is a Control Voltage (CV) input, for voltage controlling channel Pan from other devices.**

Auxiliary (AUX) Send

- **The Send connectors are used to connect to the inputs of effect devices.**
 To connect the send to a mono-input device, use the Left (Mono) output.

When a Send is connected to an effects device, the corresponding AUX Send knob determines the level of the signal sent to the effect device for each channel.

The Send Output can be taken Pre or Post channel level by using the switch to the left of the Send connectors.

Auxiliary (AUX) Return

These are normally connected to the left and right outputs of effect devices.

Master Left/Right Outputs

- **The Master outputs are auto-routed to the first available output pair in the I/O device.**
 If used in a Combinator, the Master Outs are normally connected to the "From Devices" connectors on the Combinator.

Chapter 52
Working with Players



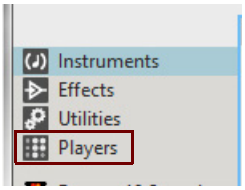
About this chapter

This chapter describes the procedures and techniques for working with the Players device type.

Overview

A Player is a special type of device that automatically processes, filters and generates MIDI Notes, based on input MIDI Notes to an Instrument device in the rack. Players can also play back MIDI on their own, without any MIDI input; this could for example be pattern sequencers.

The Player devices can be found in the Players palette below Utilities in the Reason Browser:



The Players palette in the Browser.

The basic idea behind Players is that you first create an Instrument device, then hook up one or more Player devices to the Instrument device. If the Player device is a chord generator, for example, you can have it generate chords in the desired key and scale by just playing single notes on your MIDI Control Keyboard. Absolutely great for inspiration!

Player devices are a little narrower than regular devices, and are always created directly above their corresponding Instrument device(s):



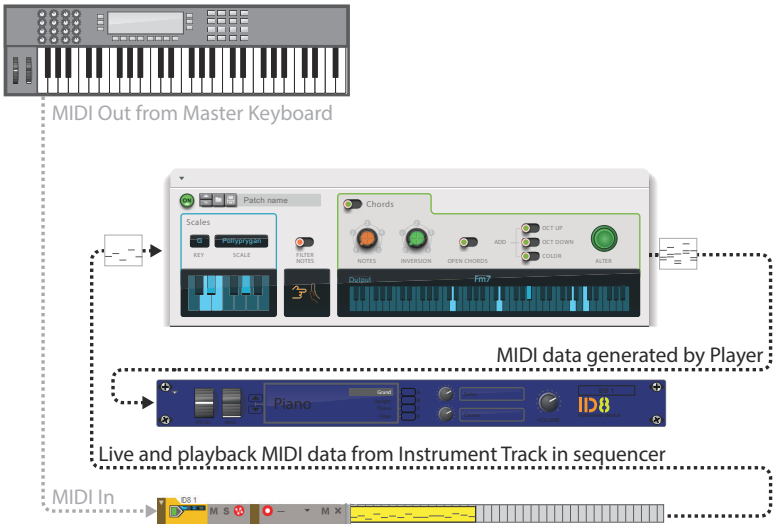
Two Player devices (in series) attached to an ID8 Instrument device in the rack.

General recording principle

Apart from sequencer Players, recording Instrument tracks using Players is done as follows:

- **By recording the notes exactly as you play them on your MIDI Control Keyboard.**
 This means that you simply record the notes you play on your MIDI keyboard straight onto the sequencer track, just as you would a regular MIDI track (without any Player). During playback the recorded notes are then run through the Player, which generates the Player notes in real-time.

The pictures below illustrates the recording principle:



The recording principle.

Using Players

Creating Players

Creating a Player device is very similar to creating other device types, with a few exceptions:

- **A Player device can not exist on its own, without an associated instrument device.**
- **A Player device can only be created for devices in the Instrument palette.**
 All Instrument types are supported: native devices as well as Rack Extensions.

Here are the different ways you can create a Player device:

→ **Drag a Player from the Players palette and drop above an Instrument device, to attach it to the Instrument device.**
The insertion point is indicated by the standard orange insertion line with the + symbol:

→ **Select the Instrument device you want to attach a Player to in the rack, and then double-click the Player device in the Players palette.**
The Player device is attached to the Instrument device.

→ **Right-click an Instrument device in the rack and select "Create > Players > Player device" from the context menu.**
The Player device is attached to the Instrument device.

- **If you only create a Player device, a MIDI Out Device will be automatically created and attached to it.**
 You could then replace the MIDI Out Device with another instrument device if you like.

Chaining Players

If you like, you can attach several Player devices to a single Instrument device in the rack. Just follow the instructions in "Creating Players" above and make sure you insert the additional Player(s) somewhere in between the Instrument device and the top section of the topmost Player device.

! **When you chain several Player devices the MIDI is processed from the top device and down.**

See "Tips & Tricks" for some interesting combinations!

Re-ordering chained Players

You can also rearrange the order of the chained Players:

→ **Drag the desired Player and drop at the new position.**
The insertion point is indicated by the standard orange insertion line with the + symbol.

Replacing Players

Replacing a Player device with another Player device can be done in exactly the same way as when replacing other device types, see "Replacing devices".

Deleting Players

Deleting Players is done in exactly the same way as when deleting other device types, see "Deleting devices".

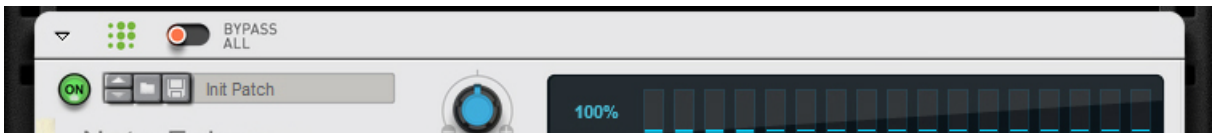
Naming Players

Naming Players is done in exactly the same way as when naming other device types, see "Naming devices".

About Players in Combinators

Players can also live inside Combinators, if there are instrument devices there. A Combinator device can also have Players attached to it (outside of it).

Common Player device parameters



Fold and Bypass All

At the top of the Player device - or above the topmost Player device, if you are using several chained Players - are the Fold and Bypass All buttons.

→ **Click the triangular Fold button to fold/unfold all Player devices attached to the Instrument device.**

→ **Switch on Bypass All to send the MIDI from your MIDI Control Keyboard straight to the instrument device, bypassing any Player devices.**

On button

→ **Click to activate or deactivate the Player.**
When deactivated (off), the MIDI notes from your MIDI Control Keyboard bypass the Player.

Patch section

Here is where you can browse, load and save your Player patches. The functionality is exactly the same as for other patch based devices, see "About patches".

Inputs and Outputs

Player devices can also feature CV In/Out connectors as well as Audio In connectors on the rear panel.

Getting the Player MIDI output onto a track in your DAW

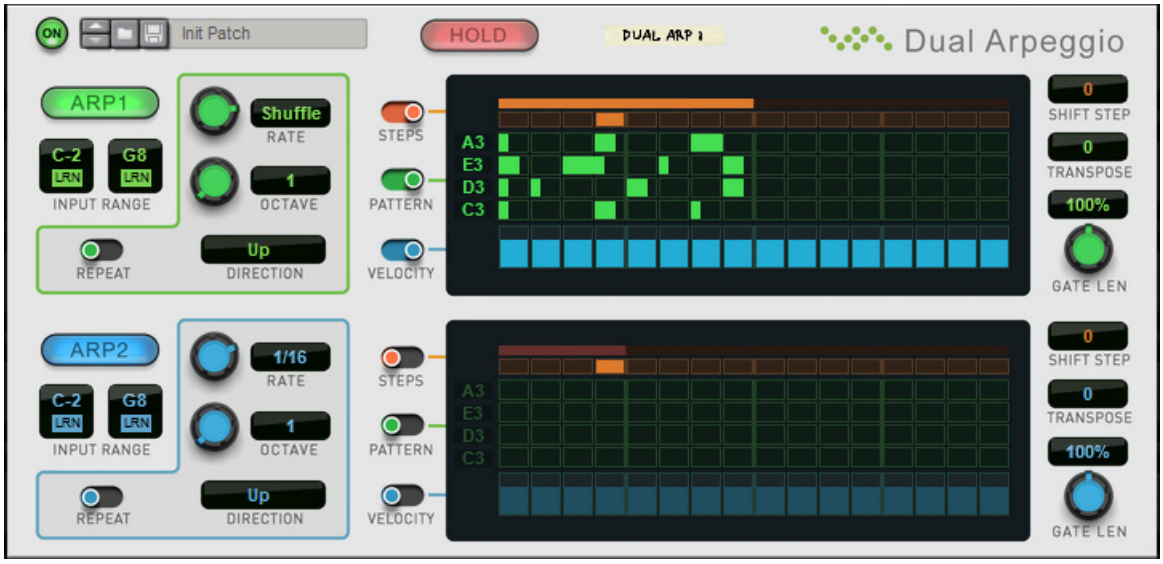
To get the generated MIDI notes onto a track in your DAW you will have to use the MIDI Out Device:

1. **Drag a MIDI Out Device from the Instruments palette in the Browser and drop in the rack.**
 The MIDI Out Device is created in the rack.
2. **Click the Players palette and drag e.g. a Dual Arpeggio Player and drop above the MIDI Out Device:**



 The Dual Arpeggio Player is automatically attached to the MIDI Out Device.
3. **Create a MIDI track in the DAW sequencer.**
4. **Select Reason Rack Plugin as MIDI Input port for that track (refer to the DAW manual).**
5. **Arm the destination track in your DAW for recording and record the Player MIDI in real-time.**

Dual Arpeggio



The Dual Arpeggio device accepts one or several notes as input and generates rhythmic patterns based on these notes. You can use it as a traditional "monophonic" arpeggiator, where arpeggio lines are played back one note at a time, based on the notes you play/hold on the MIDI Control Keyboard.

You can also use it in Pattern Mode, where you can create polyphonic (up to 4 notes) rhythmic patterns, depending on how many notes you play/hold on your MIDI Control Keyboard.

You can control the Velocity of the arpeggiated notes from your MIDI Control Keyboard, or by activating the Velocity switch and drawing in Velocity values in the display.

Dual Arpeggio has two independent arpeggiator sections, that are identical. The sections can either be played by the same input notes (parallel) or you can set up separate key ranges for them (key split). Input notes that are outside the set range(s) will be throughput as is, which means you can play melodies on top of arpeggiated chords etc.

The Dual Arpeggio device plays back in sync with the main sequencer, as soon as you input notes to it.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

Arp1/Arp2

These are the "on/off" buttons for each of the sections.

Input Range

Here you can define the input note range, to which the arpeggiator should respond. Input notes outside the set range(s) are bypassed and let through unaffected. This is very useful if you, for example, want to play melodies on top of arpeggiated chords.

→ **Set the lowest and highest input note that should be routed to the corresponding arpeggio section, by click-holding the notes in the displays and dragging up/down.**

You can also set the Input Range by using the LRN (Learn) buttons:

→ **Press a key on your MIDI Control Keyboard and then click the desired LRN button to input the note number for the pressed key.**
Repeat the procedure in the other display for the other note in the range.

Rate

→ **Set the time division to which to play back the arpeggiated notes.**
Values: 8 bars, 4 bars, 3 bars, 2 bars, 7/4, 6/4, 5/4, 4/4, 3/4, 2/4, 3/8, 1/4, 3/16, 1/8, 1/8T, 1/16, 1/16 Shuffle*, 1/16T, 1/32, 1/32T, 1/64.

* The Shuffle amount is set with the Shuffle knob in the I/O device.

Octave

→ **Set how many octaves the arpeggiated notes should cover.**
Range: 1-4 octaves.

Provided the Pattern function (see "Pattern") is not active, this is what will be played back:

Octave range	Description
1 Oct	The arpeggiated notes will be those that you press down on the keyboard.
2 Oct	If you select this, the arpeggio will be extended to a two octave range, i.e, the arpeggio pattern is played then repeated one octave up. In Random mode, the notes you input are played in random order over 2 octaves.
3 Oct	Same as 2 Oct but extended to a three octave range.
4 Oct	Same as 2 Oct but extended to a four octave range.

Repeat

→ **Activate this to repeat each note in the generated arpeggio.**
Since each note is played back twice at the set Rate, the total length of the arpeggio will be twice as long.

Direction

→ **Click-hold in the display and drag up/down to select the direction of the arpeggio/pattern notes.**
Values: Up, Down, Up+Down and Random.

Provided the Pattern function (see "Pattern") is not active, this is how it works:

Mode	Description
Up	This will generate an arpeggio that plays from the lowest note to the highest note.
Down	Notes are played from the highest note to the lowest note.
Up+Down	Notes are played from lowest note to highest, then from highest back down to the lowest. The very lowest and the highest arpeggio notes are not repeated. I.e. the arpeggiator will play the lowest note to the second highest note, then the highest note to the second lowest note.
Random	The notes you input will be arpeggiated randomly.

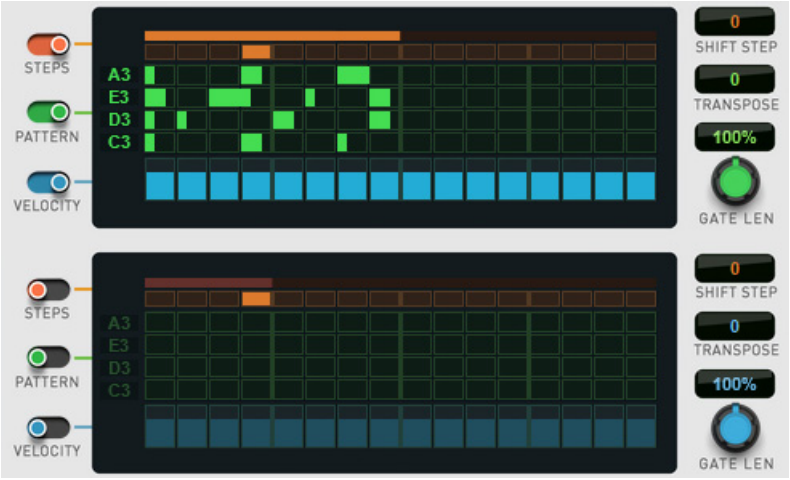
If the Pattern function is active (see "Pattern"), the Direction setting only determines in which direction the steps should advance - regardless of the pitches of the notes in the pattern.

Hold

→ **Click the Hold button to keep the arpeggio running even after you release all keys.**
It will continue to arpeggiate the last notes played until a new note-on is received.

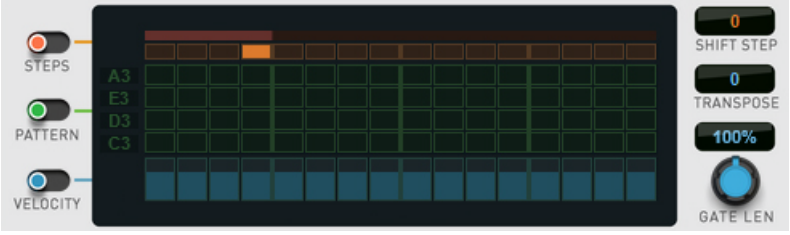
- **If you continue to hold down at least one key when Hold is on, any new notes will be added to the existing arpeggio as opposed to starting a new arpeggio.**

The Display sections



In the display sections you can design your arpeggios/patterns.

Normal (Arpeggio) mode



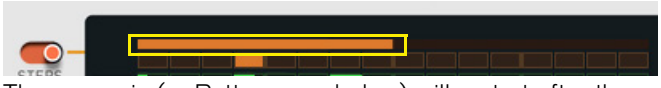
If none of the switches to the left of the display (Steps, Pattern or Velocity) are activated, the display only shows the orange moving step indicator (if you hold down more than one key on your MIDI Control Keyboard). The rest of the display area is dark.

- **The step indicator travels the same number of steps as the number of notes you play and hold on your MIDI Control Keyboard.**
 So, if you hold down a 5-note chord, the step indicator advances five steps before it starts over again.

Steps

If you want the arpeggio to always play a certain number of steps, regardless of how many keys you hold down, you can activate the Steps function. This is perfect for maintaining a steady “beat” in your song.

1. **Click the Steps switch to activate the Steps function.**
 The default value is 4 steps.
2. **Click or click-hold and drag on the right side of the orange bar on the area above the step indicator, to increase or decrease the number of steps:**



The arpeggio (or Pattern, see below) will restart after the set number of steps, regardless of how many notes you are playing.

Range: 1-16 steps.

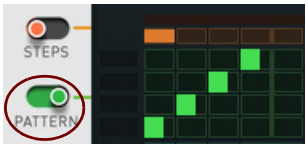
Pattern

In the default “Arpeggio mode” (Pattern switch off), the arpeggiator(s) work in the standard “monophonic” fashion, where the notes are played back one by one according to the settings you have made in the left section of the device.

With Pattern activated, the arpeggiator turns into a combined pattern player + arpeggiator. Pattern mode allows for up to four-note polyphonic “arpeggios”.

Here is how it works:

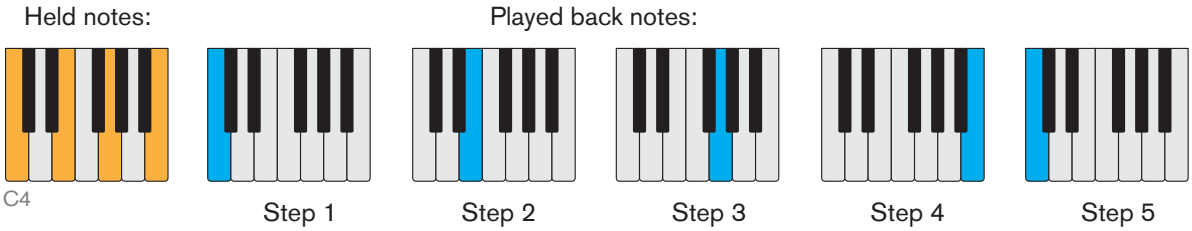
1. **Click the Pattern switch to activate the Pattern function.**



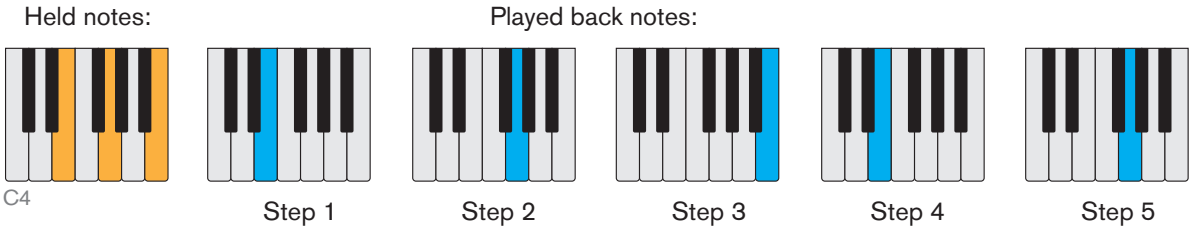
All currently active pattern steps (green boxes) light up in the display.

The four green boxes in the example above represent the playback pattern if you hold a 4-note chord on your MIDI Control Keyboard. The lowest row in the display represents the lowest held note in the chord you play.

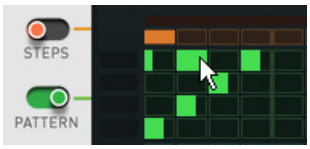
2. **Set the Direction to “Up” (see “Direction”), to make the examples below work as described.**
If you, for example, hold down the keys C4, E4, G4 and B4, the pattern plays back C4 in step 1, E4 in step 2, G4 in step 3 and B4 in step 4. Then the pattern starts over again:



If you hold down fewer keys, the pattern steps gets equally fewer steps and only plays back the new lowest to the new highest notes in the chord:



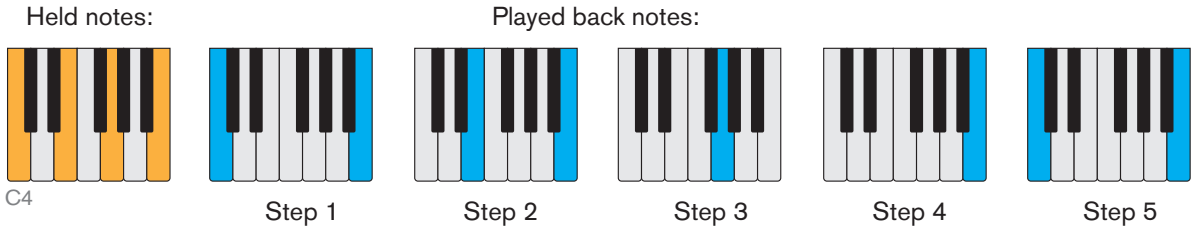
3. **Click in any of the matrix boxes to edit the pattern:**



Editing the pattern.

- **Each step can have a note that can be 1/3, 2/3 or 1/1 of the step's total length, depending on where in the box you click.**
 This way you could draw in "staccato" notes for desired steps, if you like. If you click the full 1/1 of a step (click in the right area of the box), and then click another consecutive box on the same row, the notes will be tied together (played in legato).

 If you hold down the keys C4, E4, G4 and B4, the pattern above now plays back C4 and B4 in step 1, E4 and B4 in step 2, G4 in step 3 and B4 in step 4. Then the pattern starts over again:



→ **Click anywhere on a filled box to delete the note.**

→ **To fill several consecutive boxes on the same row with identical note lengths, click and hold on the desired position in the initial box and then drag to the right:**

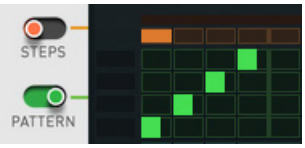


Adding consecutive notes by click-holding and dragging.

Here are some general rules regarding Pattern Mode:

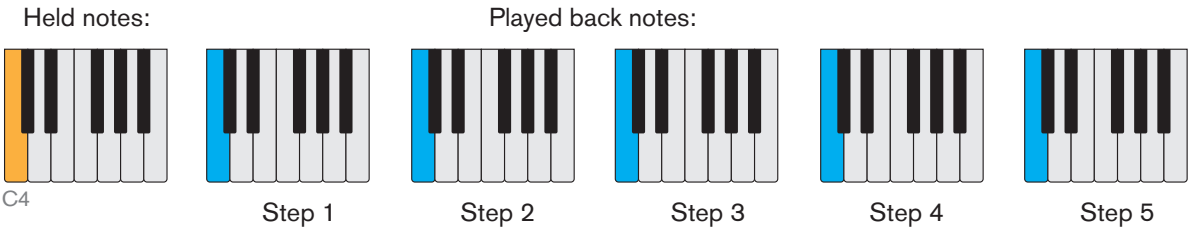
- **A pattern can be up to 16 steps long (if Steps is activated, see "Steps").**
- **The notes you play are then mapped to the four note positions for each step.**
 If you play a four note chord, some steps may play only the lowest note, others may play a higher note or a combination of notes or all four.
- **If you hold down more than four notes, note 5 and up are ignored completely in the pattern.**
 Note that if the Steps function is not active (see "Steps"), note 5 and up will add additional silent steps in the pattern, making the pattern longer.

- **If you play fewer notes than the number of rows you have programmed in you pattern, this is what happens:** Let's take the pattern below (with the Steps function off) as an example:

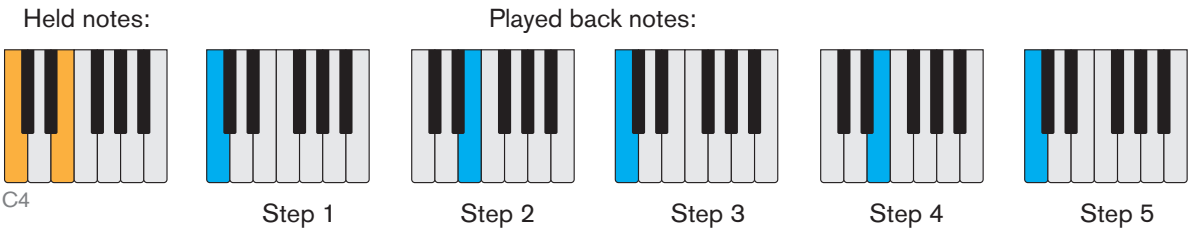


Pattern Mode with the Steps function off.

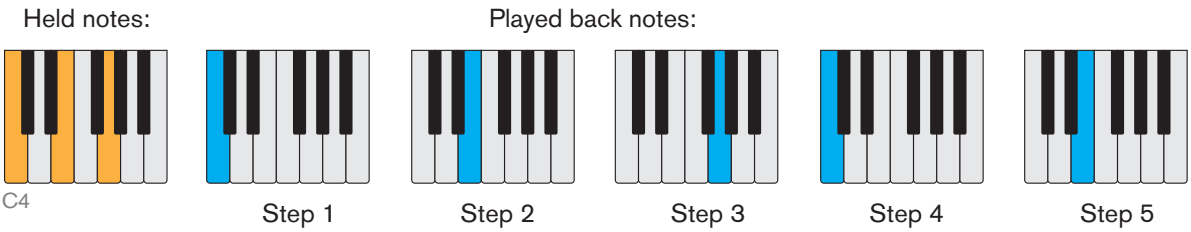
If you only hold down key C4, note C4 is repeated (since the pattern length is only one step with one key pressed):



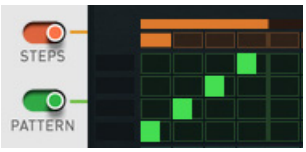
If you hold down C4 and E4, the pattern plays back C4 and E4 alternating (since the pattern length is now two steps with two keys held):



If you hold down C4, E4, and G4 the pattern plays back C4, E4 and G4 alternating (since the pattern length is now three steps with three keys held):

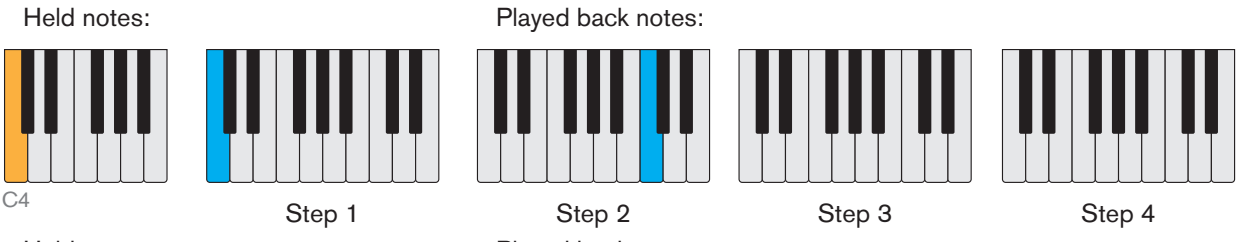


- **If we now do the same procedure with the Steps function on - and set the number of steps to 4 - things work a little differently.**
 This is what happens:

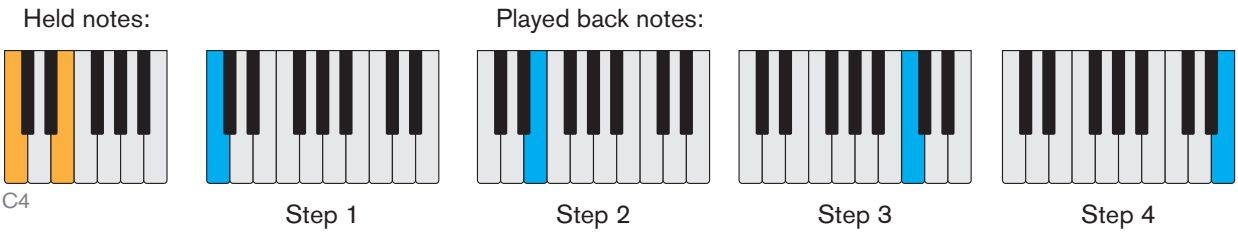


Pattern Mode with the Steps function on.

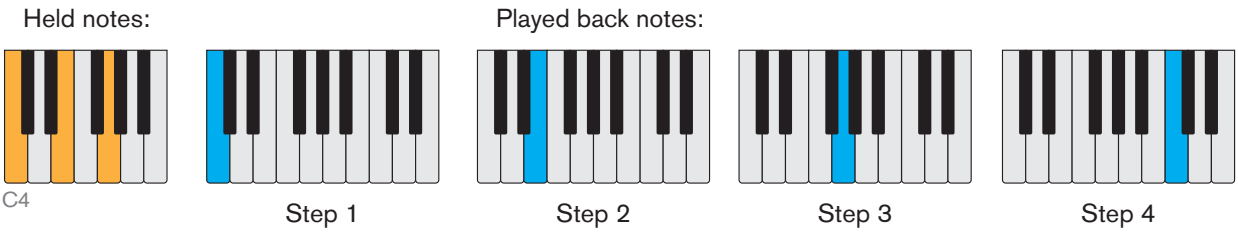
If you only hold down key C4, note C4 is played back on step 1 and note C5 is played back on step 2. Steps 3 and 4 are silent. In other words, a new note one octave above is automatically generated on the second step:



If you hold down C4 and E4, the pattern plays back note C4 on step 1, note E4 on step 2, note C5 on step 3 and note E5 on step 4. So, the two held notes are played back one octave higher on steps 3 and 4:



If you hold down C4, E4, and G4 the pattern plays back C4 on step 1, E4 on step 2, G4 on step 3 and C5 on step 4. As you can see, the lowest held note is played back one octave higher on step 4:



Velocity

In default mode, when the Velocity function is off, the velocities of the input notes set the velocity of each generated note in the arpeggio/pattern. If you want to define the velocity values and draw them in manually, you can activate the Velocity function.

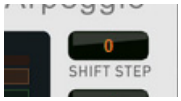
1. **Click the Velocity switch:**



2. **Click or click-hold and drag the blue Velocity bars to manually set the velocity values.**
 When you play your chords, the input velocity is disregarded and replaced by the values you drew in the display.

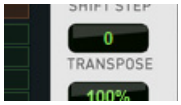
 Range: 0-127.

Shift Step



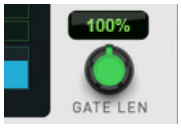
- **Change where in the arpeggio/pattern you want the playback to begin.**
 Range: -16 to + 16 steps.

Transpose



- **Transpose all the generated arpeggio/pattern notes equally, in semitone steps.**
 Range: -24 to + 24 semitones.

Gate Length



- **Change the lengths of the generated arpeggio/pattern notes.**
 Here you can change the “gate/hold” time for the generated arpeggio/pattern notes. 100% means the gate time is exactly one full Step. A Length of 25% results in a gate time of 1/4 of a Step and a Length of 400% means a gate time of four Steps.

 Range: 25%-400%.

Note Echo



Note Echo simulates a MIDI delay effect, by adding repeats of the incoming MIDI notes. The Note Echo device is polyphonic, so you can get repeats of entire chords if you like.

! **Please, note that this device is not available in Reason Lite Rack Plugin.**

Step Length

The Step Length knob controls the echo time, and consequently also the time between the repeats. The step length range is 0 to 1000 milliseconds. When Tempo Sync (see below) is enabled, the range is 1/128 notes up to ½ note.

► **A Step Length of 0 makes it possible to create chords and clusters, since all steps are played back simultaneously. See "Creating parallel chords" for a practical example.**

Tempo Sync

Switch on to sync the Step Length (see above) to the main sequencer tempo.

Repeats

This knob sets the number of repeats, from 1-16. The number of repeats is also shown in the display.

Velocity

The Velocity knob controls the linear increase or decrease of the Note Velocity value for each step.

The range is from 10% to 200%, where 100% means the velocity value is left unchanged throughout the repeats. Values below 100% means a decrease of the velocity values, and values above 100% means increasing velocity with each repeat.

The velocity values are represented by blue bars in the display. The blue line in the bars indicates the 100% value.

Pitch

With the Pitch knob you set how many semitones each step should be transposed relative to the previous step. The function is linear for all repeats. The Pitch values are represented by orange bars in the display.

The range is −12 to +12 semitones. 0 is the default value and means no transposition of the input notes.

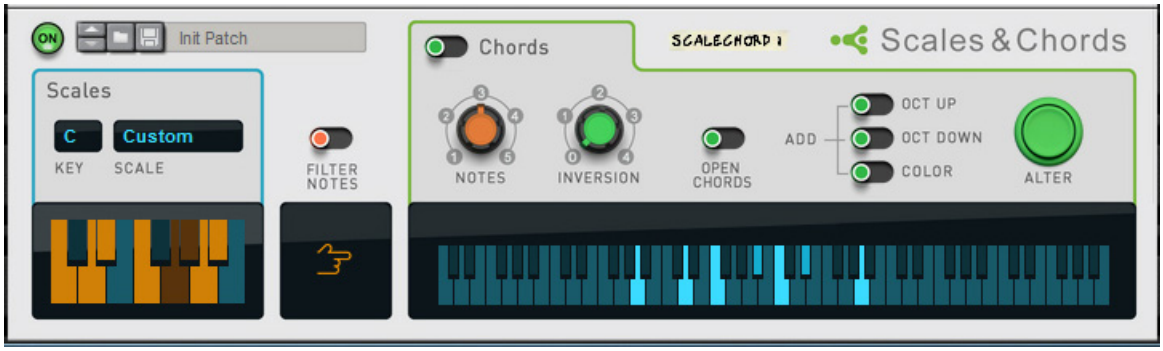
! **If you use very high or very low Pitch value settings together with many Repeats, the note range of the associated instrument might be exceeded. If this happens, any remaining repeats will be silent.**

Display

Aside from showing number of Repeats, Velocity and Pitch (transposition) information, the display can also be used if you want to mute individual repeats.

→ **Click the desired green circles in the display, to mute the corresponding repeat.**
This way you can create interesting rhythmic repeats. The first green circle (to the left in the display) represents the incoming "dry" MIDI Note. This can be muted as well, if you like. By default, all repeats are on (unmuted).

Scales & Chords



The Scales & Chords device either transposes incoming notes to fit a set scale, or transposes notes and generates chords that fit the desired scale. Scales & Chords has a number of built-in preset scales, and you can also create your own custom scales!

Here are some examples of how you can use it:

- **to get chords from single note inputs.**
- **to get new chord voicings and progressions that feel fresh and inspires your music making.**
- **to experiment with existing note lines, by transforming them to different scales.**
- **to get the notes you play to automatically fit the music.**
- **to easily be able to play chords without using a MIDI Control Keyboard.**
 You could write single MIDI notes in a Note clip and have the Scales & Chords Player generate the chords for you.
- **to get chords that sound right, without knowledge of tonality/scales.**

Scales

In the Scales section you choose which key and scale you want.

- **Note that all parameters in the Scales section can be automated, so you could change key and scale anywhere throughout your song if you like!**

Key

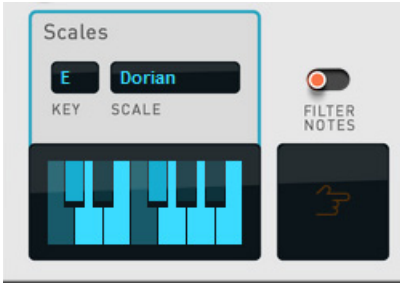
- **Click to select the desired Key from the list that appears.**
 12 Keys are available, from C to B.

Scale

- **Click to select the desired Scale from the list that appears.**
 13 scales are available, plus a Custom scale that you can create yourself (see below). The preset Scales are:

 Major, Minor, Lydian, Mixolydian, Spanish, Dorian, Phrygian, Harmonic Minor, Melodic Minor, Major Pentatonic, Minor Pentatonic, Hemi Pentatonic and Chromatic.

- **The notes that fit the selected Key and Scale are displayed in light blue in the keyboard display in the Scales section:**



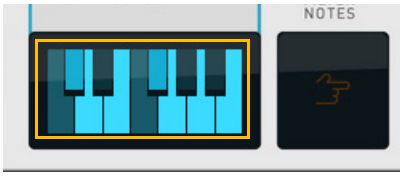
The Dorian scale in the key of E.

The notes in the selected Key and Scale are then automatically repeated throughout all octaves of the entire MIDI Note range.

- **Notes you play on your MIDI Control Keyboard/On-screen Piano Keys are indicated in orange in the Scales keyboard display.**

Creating a Custom Scale

→ **Create a custom Scale by clicking the desired keys in the keyboard display in the Scale section:**



Selected notes are colored light blue. The selected notes are then automatically repeated throughout all octaves of the entire MIDI Note range.

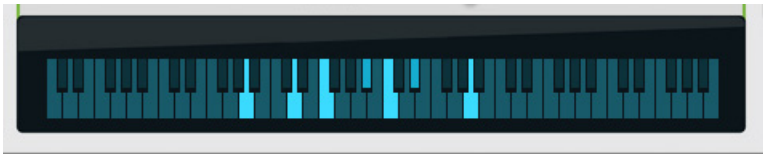
- **You can easily switch between your Custom Scale and any of the preset Scales whenever you want, by clicking the "Scale" display and selecting the desired scale from the list that appears.**
- **Your Custom Scale is automatically saved with the song.**
 Your Custom Scale is also saved in the Scales & Chords patch, should you choose to save your settings as a patch.

Filter Notes

- **When the Filter Notes switch is off (default), incoming notes outside the set Scale (see "Scales") are transposed to fit the scale.**
 Wrong notes are automatically transposed to the closest correct note. If the wrong note is equally near two correct notes, the wrong note is transposed to the lowest of the correct notes.
- **When the Filter Notes switch is on, incoming notes that are outside the set scale will be filtered out (silent) instead of transposed.**
 The display below shows a hand symbol whenever wrong input notes are filtered out (silent).

Chords

→ **Switch on the Chords function to generate chords out of single notes.**
When you play a note, you will get a chord that fits the set Scale (see "Scales"), and contains the number of notes set with the "Notes" knob. The generated notes are shown in real-time in the keyboard display at the bottom of the Chords section:



! **If you play several notes at the same time you will get several chords, which might be cool or just weird. If notes in the generated chords should coincide, only one instance of the notes are played back (to avoid doubled/layered notes).**

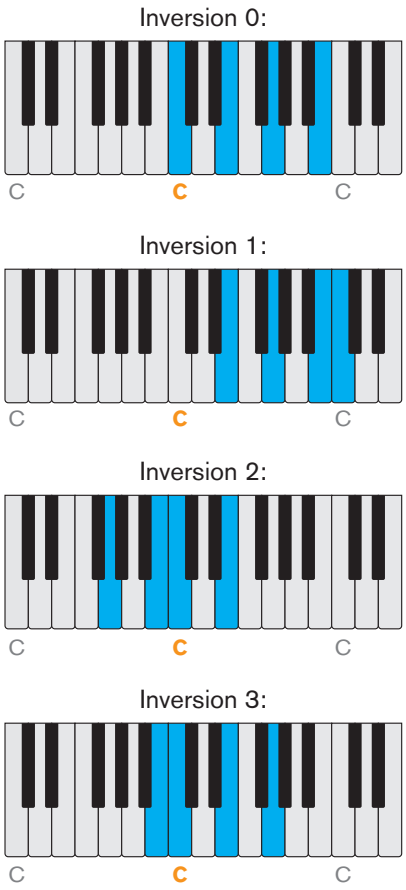
► **If you want to create "parallel chords", i.e. chords that contain notes with the same relative intervals, the Note Echo device is better suited for this, see "Creating parallel chords".**

Notes

→ **Set the desired number of notes in the generated chords.**
Range: 1-5 notes.

Inversion

The Inversion knob defines which note in the generated chord should be the lowest. The function tries to keep the lowest generated note in the chord as close to the played root note as possible. Below is an example of inversions of a generated 4-note chord in the Scale C Major, played from a C note of the MIDI Control Keyboard:



Inversions of a generated 4-note chord played from the (orange) C note.

! **Note that the Inversion function is only relevant if the Inversion number is less than the selected "Notes" value. So, if the Notes value is 4, only a maximum Inversion value of 3 is useful.**

Open Chords

When the Open Chords function is active, some notes in the generated chords are transposed in octave steps, so that the notes are not so close to each other. The transposition is different depending on the "Inversion" and the "Notes" settings.

Add

The Add functions can be used for adding notes to the generated chords:

- **The Add Oct Up function adds a note one octave above the root note.**
- **The Add Oct Down function adds a note one octave below the root note.**

- **The Add Color function adds a note two thirds above the highest note in the tertian chord structure.**
 If the Notes value in the Chord section is 3, you will get a regular triad (e.g. a C major). Adding a sixth above top will mean adding a 9 (e.g. the note D). Similarly, if the Notes value in the Chord section is 4, this will add an 11; if the Notes value in the Chord section is 5, this will add a 13.

► **All three Add functions can be used together, simultaneously.**

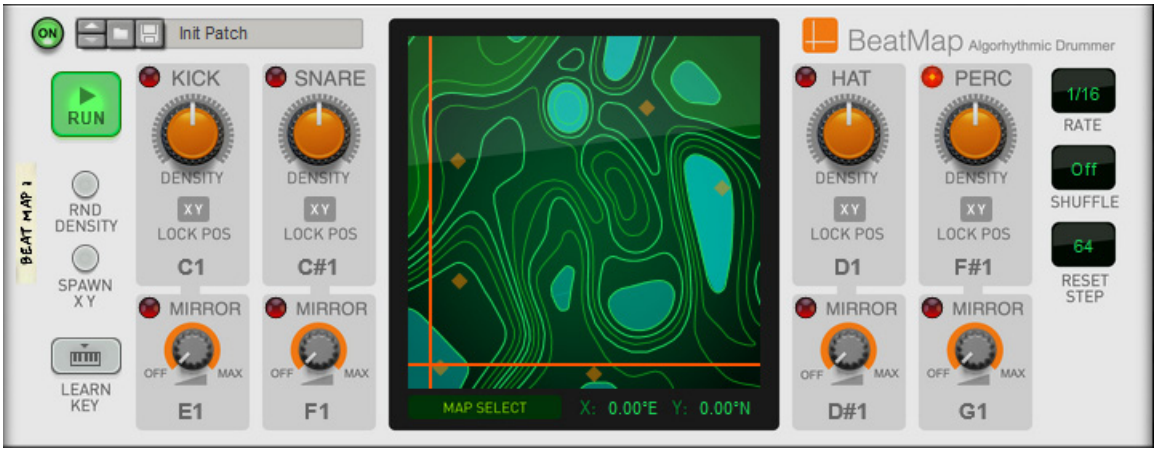
! **If the added note is already used in the chord (due to inversions, open chord etc), the note is transposed up another octave (or down, in case of Oct Down).**

Alter

The Alter button is a momentary button that turns off as soon as you release the mouse button (like the Trig button on The Echo, for example). It alters the chord slightly so that it goes outside the current scale. The result depends on the Key (as set in the Scale section), whether the current scale is major or minor and what note you play (relative to the Scale Key).

The chord is altered by having one of the chord notes changed. For example, a major chord may be changed to a minor chord, or vice versa.

Beat Map

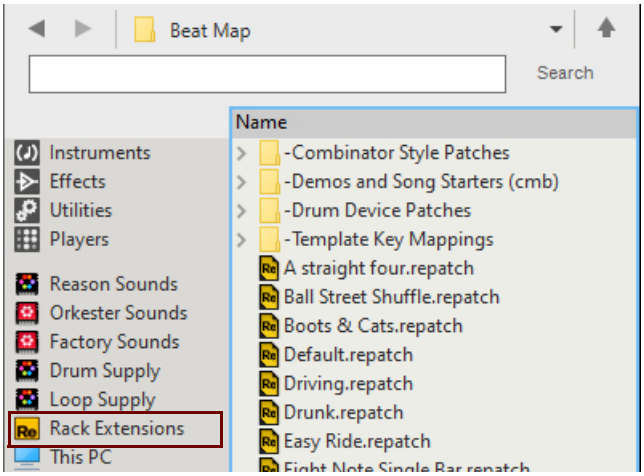


The Beat Map Algorhythmic Drummer is a Player that generates drum patterns based on built-in beats, algorithms and simple but powerful controls. It's normally used with a drum instrument such as Kong, Rytmik or Umpf, but can also be used with melodic instruments for interesting results.

! **Please, note that this device is not available in Reason Intro Rack Plugin or in Reason Lite Rack Plugin.**

Included content

To browse the content of Beat Map, click "Rack Extensions" in the Browser and then double click the Beat Map folder:



Beat Map comes with a number of patches, both for the Player itself and for several different drum devices. The drum device patches are tailored to work well with Beat Map, but you can of course use any patch from the Factory Sound Bank, or make your own.

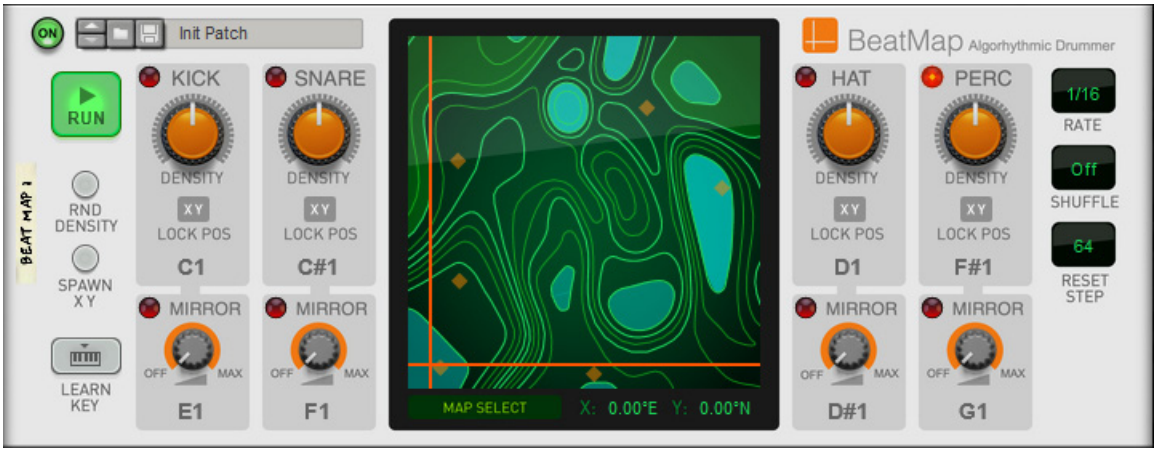
If you want immediate results, start by loading one of the patches in the Combinator Style Patches folder! These combine Beat Map with drum devices and effects for instant rhythmic goodness.

If you want a taste of what can be done with multiple Beat Maps in combination with other devices, check out the Demos and Song Starters folder.

! **Note that some of the combi patches use Rack Extensions included with Reason Suite.**

Finally, the Template Key Mappings folder contains Beat Map patches where the keys (note numbers) have been adapted for other drum configurations such as GM, Reason Drum Kits and various 3rd party drummers.

The front panel



The central item on Beat Map's panel is the Map display. This is where you select a basic rhythm, by clicking and dragging the cross-hair.

- **The cross-hair position is also indicated with the X and Y position parameters, which you can edit directly if you like.**

Each integer XY position corresponds to a different beat. The map graphics are there for visual reference and for the nice looks – there is no particular rhythmic significance to the elements on the map. However, as a general rule, moving the position to the right will make the bass drum pattern more syncopated and moving the position upwards will make the snare pattern more syncopated.

- **Clicking the Spawn XY button drops you at a random position:**



Map Select

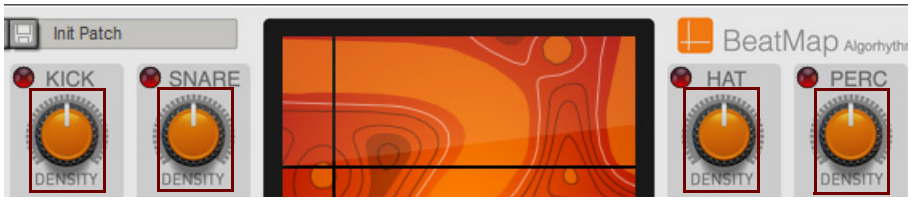
1. **Click the Map Select button to show the four included maps:**



They contain beats in different styles – click on a map to see some info about it in the display.

2. **To select a map for use, either double click it or select it and click Map Select again.**

Density

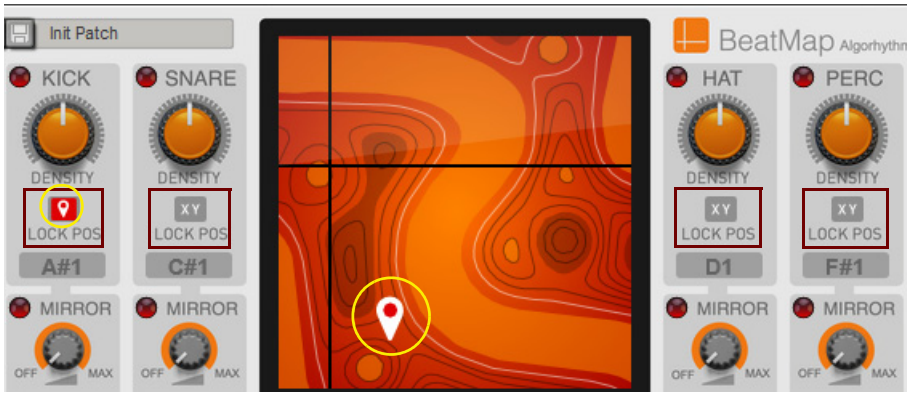


The beats consists of four different rhythmic patterns, for kick, snare, hi-hat and percussion. Once you've found a beat that you like, you can adjust the Density of each drum. This means reducing or increasing the number of notes. Turning Density down to zero will mute that drum. The Density parameters can be automated or CV controlled, for continuous variations.

- **Clicking the Random Density button to the left will change all four Density controls to random values, useful for live variations:**



Lock Position

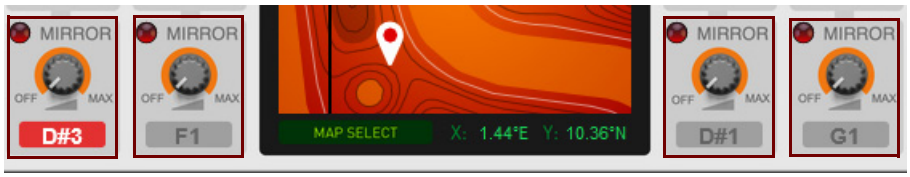


You may find a beat where for example the kick is perfect, but not the other drums. Then you can click Lock Pos for the kick. This will put a "pin" on the map – the kick beat will stay at this position but you can continue exploring the map for the other instruments.

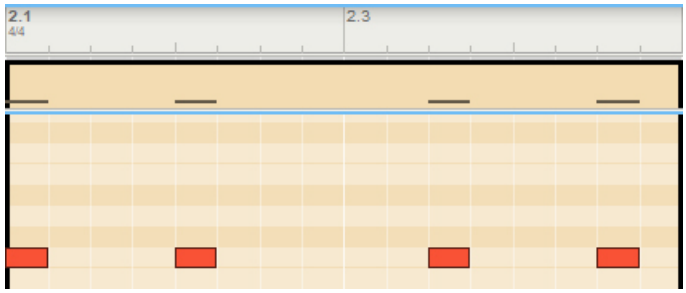
This can be done independently for all four drums, locking them to different map positions. This way you can combine rhythms from different basic beats, for infinite variations.

! **You can still adjust Density for locked drums.**

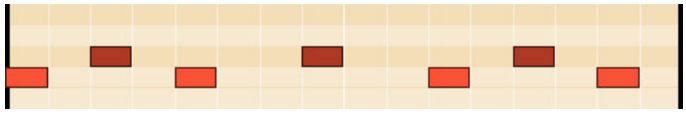
Mirror notes



Each of the four drums has a "Mirror" function, which plays a note in between two note positions from it's main drum. For example, if the kick beat plays like this:



... then its Mirror notes (dark red) will play like this:



This means that the Mirror notes will be affected by the Density of the main drum, creating interesting counter-rhythms or ghost notes.

Use the Mirror knobs to set the velocity (typically level) of the Mirror notes. Turning the knob down completely will turn off the Mirror note.

Setting MIDI note numbers

Each drum and Mirror function sends out notes on a separate MIDI note number, meaning that Beat Map can control eight different drum sounds.

- **Clicking the red signal LED for a drum will send out that note, for quick auditioning of drum sounds:**



The Perc drum sends out MIDI Note F#1

- **You can set which note a drum should send by clicking and dragging the note number on the panel:**



Alternatively, use the Learn function:

1. **Click Learn Key:**



2. **Click the note number value for a drum or Mirror to highlight it.**



3. **Play the desired note on your MIDI controller.**
4. **Repeat for the other notes.**
5. **Click the Learn Key again to exit Learn mode.**
 This is required for the changes to stick when you save the patch or song.

Global settings



To the right are three settings that affect the overall playback:

- **Rate is the note value of each step. Normally this is set to 1/16th notes.**
- **Shuffle allows you to add a swing feel to the beats by delaying the offbeat 1/16th notes.**
 You can set it manually from 50-75% or select “Global” to use the Global Shuffle setting (found in the I/O device in Reason Rack Plugin).
- **Reset Step determines the number of steps before the beat starts over from the beginning.**
 The default is 64, but it can be useful to lower this to make a beat less varying or to impose some structure on a chaotic rhythm.

Beat Map and the main sequencer

You can click the Run button manually for auditioning the beat, but the Run function is normally controlled by the Play function in the main sequencer.

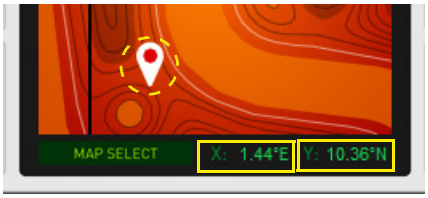
However, if you want Beat Map to play in certain parts of your song but not in others, it's possible to control this by automating the Beat Map “On” button in your host DAW:



In addition, most Beat Map parameters can be automated. For example, you can record map movements or Density changes.

A couple of special features:

- **The Locked XY positions can be automated for each drum.**
 This allows you to lock a single drum to a rhythm but change it with automation:

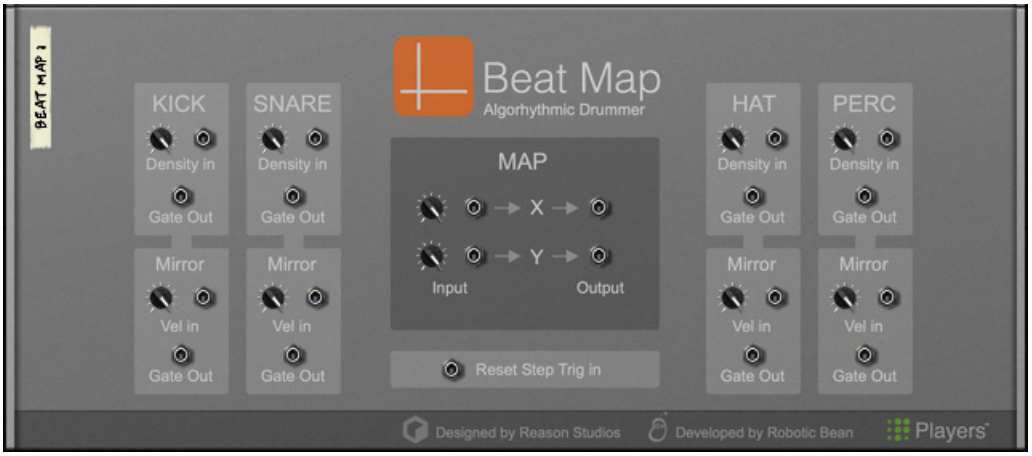


- **The Map selection ("Beat Map" parameter) can be automated or controlled from a Combinator.**
 Switching to another map can be useful as a break or fill.

Editing the drum notes

To manually edit the beats from Beat Map you need to connect Beat Map to a MIDI Out Device and record the MIDI output from Reason Rack Plugin to another track in your DAW, see "Getting the Player MIDI output onto a track in your DAW".

Using CV



On the back of Beat Map, you'll find a comprehensive selection of CV inputs and outputs.

- **Density CV in and Mirror Velocity CV in for each drum.**
 Using these with a Pulsar CV LFO can create longer, varying rhythmic patterns or random variations. There are plenty of examples of this among the Combinator Style Patches.
- **Gate Out from main drums and Mirror sections, allowing you to trigger other sounds or effects in parallel.**
- **Reset Step Trig In.**
 Whenever this receives a positive CV value, the beat will start over from the first step.
- **Map XY CV Inputs, for automatically modulating the map position during playback.**
- **Map XY CV Outputs.**
 These send out the X/Y coordinates as CV values, letting you use the Map display as an XY controller for other devices in Reason.

Tips & Tricks

Generating scale-correct arpeggios from single notes

By chaining a Scales & Chords device with a Dual Arpeggio device you will be able to generate scale-correct arpeggios by playing single notes.

1. **Create an Instrument device.**
 In this example we create an ID8.
2. **Select the ID8 device in the rack and double-click a Scales & Chords device.**
 The Scales and Chords device is automatically attached to the ID8.
3. **Select the ID8 device in the rack and double-click a Dual Arpeggio device.**
 The Dual Arpeggio device is attached between the Scales & Chords device and the ID8 device:
4. **On the Scales & Chords device, select Key and Scale, then switch on the Chords function set the Notes knob to 4.**
5. **Play single notes on your MIDI Control Keyboard and hear how four-note scale-correct arpeggios are now generated.**

Generating chord arpeggios

By chaining a Dual Arpeggio device with a Scales & Chords device you will be able to generate arpeggiated chords.

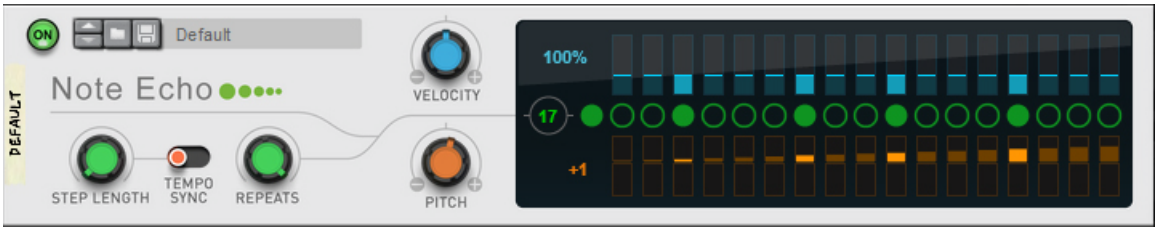
1. **Create an Instrument device.**
 In this example we create an ID8.
2. **Select the ID8 device in the rack and double-click a Dual Arpeggio device.**
 The Dual Arpeggio device is automatically attached to the ID8.
3. **Select the ID8 device in the rack and double-click a Scales & Chords device.**
 The Scales and Chords device is attached between the Dual Arpeggio device and the ID8 device:
4. **On the Scales & Chords device, select Key and Scale, then switch on the Chords function set the Notes knob to 3.**
5. **Play chords on your MIDI Control Keyboard and hear how arpeggiated three-note chords are now generated.**

Creating parallel chords

The Scales & Chords Player is great for generating chords that stay within a set scale, but sometimes you may want chords that keep their *form* when you play different notes. For example, you might want a minor 9th chord that is transposed with the notes you play, but stays a minor 9th chords (like with the Chord Memory function of some synthesizers, for example). This can be called “parallel chords”, and is actually best done with the Note Echo device:

1. **Set Step Length to 0.**
2. **Raise the Repeats value to maximum.**
3. **Set the Pitch to +1 semitone.**
 This means that each repeat is 1 semitone higher than the previous - and since Step Length is 0, they are all played at the same time. If you play a note now, you get a dense cluster of notes - not very musical or useful, but let’s continue:

4. **Click the Repeat circles in the display to turn off the notes you don't want.**
 This is how you would set up a minor 9th chord for example:



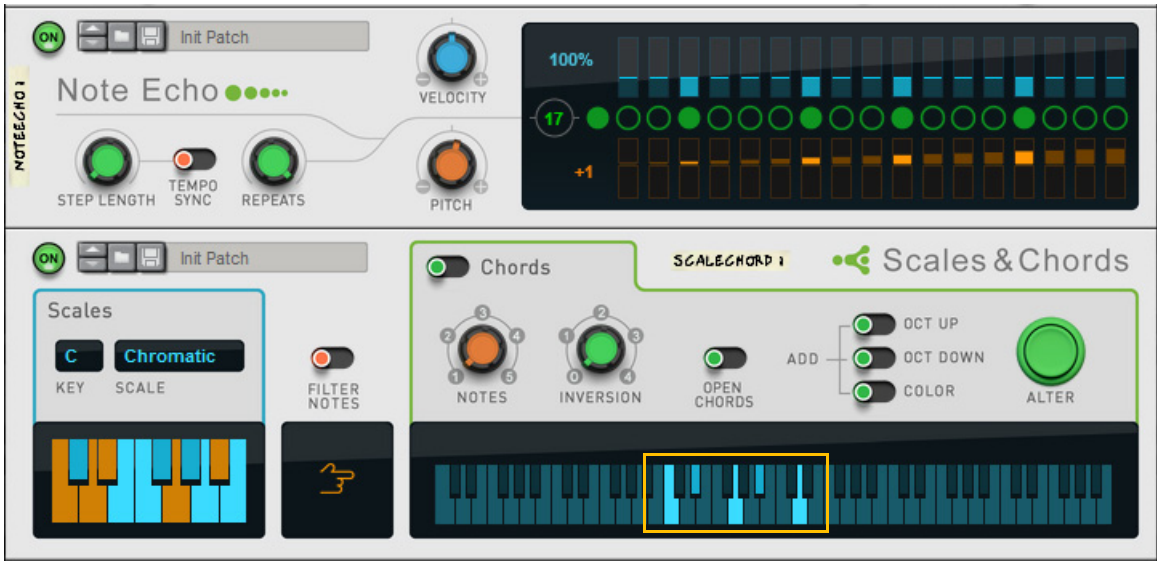
5. **Play different (single) notes on your MIDI Control Keyboard/On-screen Piano Keys to get parallel, transposed chords.**
 You could also add a Dual Arpeggio Player after the Note Echo for cool arpeggios!

► **The Note Echo device comes with a number of ready-made chord patches. You find them by clicking the Browse Patch button on the Note Echo device and opening the "Chords" sub-folder in the browser.**

Using a Scales & Chords device as a "MIDI Note monitor"

Sometimes it might be interesting to see in real-time what notes/chords a Player or "Player chain" actually generates.

1. **Add a Chords & Scales device last (at the bottom) in the Player chain.**
2. **Make sure Scale is set to "Chromatic" and that Filter Notes and Chords are Off.**
 When you play through your Player(s) the generated notes/chords are now displayed in the Scales & Chords keyboard display:

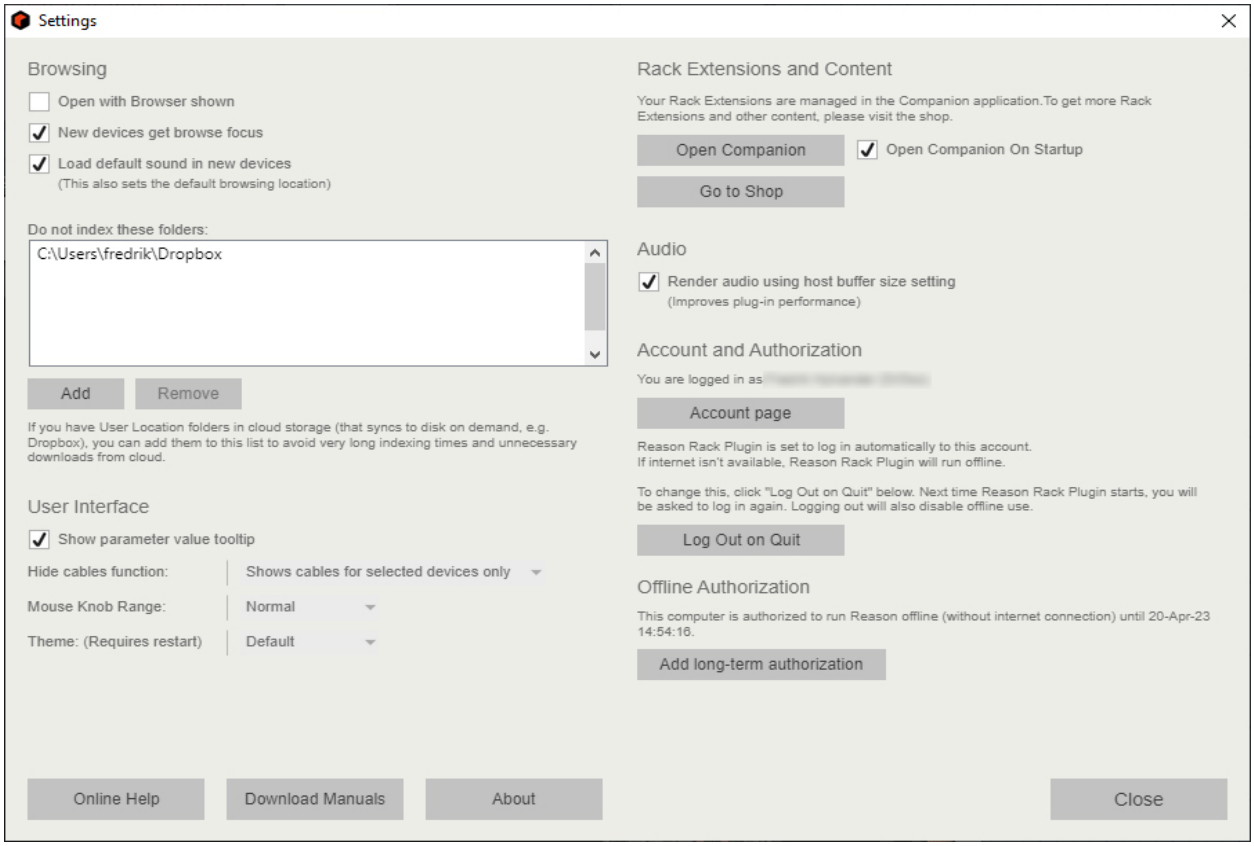


The "Creating parallel chords" example monitored in a Scales & Chords device when playing the note C.

Chapter 53
Settings

The Reason Rack Plugin Settings dialog

→ **Click Settings at the top right of the Reason Rack Plugin window to open the Settings dialog:**



Browsing

Open with Browser shown

Check this item to always show the Browser unfolded when creating a Reason Rack Plugin instance in your DAW.

New devices get browse focus

When this is active, Browse focus is automatically set to new devices as soon as they have been created. This means that you can immediately start to browse for, and load, compatible patches for the devices you just created. Browse focus is indicated by orange sidebars and an orange patch section on the devices. The top section of the Browser also gets an orange label with the device's current patch name printed on it.

Load default sound in new devices

When this is activated and you create a new device (of a device type that supports patches), a default patch will be loaded. This way you can be sure that a new device generates sound right away. The default sound will also determine the default location in the Factory Sound Bank when you browse for patches for the new device.

If you turn this off, new devices will be initialized - parameters are reset to their default values and no samples are loaded in sample-based devices.

! **Note that "Sound" also implies patches for effect devices such as the RV7000 and Scream 4.**

Do not index these folders

Here you can exclude folders that you have in the Locations and Favorites section of the Reason Browser from scanning and indexing. This is especially useful if you have cloud (e.g Dropbox) folders on your computer.

- **Click the Add button to add folders to be excluded from the indexing.**
- **Select a folder you want to remove in the list and then click the Remove button.**

User Interface

Show parameter value tool tip

Normally, if you hold the mouse pointer over a parameter on a device panel for a moment, a Tool Tip appears displaying the name and the current value of the parameter. If you uncheck this option, Tool Tips will not be displayed.

Hide cables function

The selected alternative determines how cables on the back of the rack should be displayed when "Hide Cables" is selected on the top menu in the Reason Rack Plugin window:

- **Select "Hides auto-routed cables" to only display manually routed cables.**
 All devices with auto-routed cables will have "transparent" cables to make it easier to distinguish the manually routed cables.
- **Select "Shows cables for selected devices only" to only display the cables for selected devices.**
 All other (non-selected) devices will have "transparent" cables to make it easier to distinguish the cables of selected devices.
- **Select "Hide all cables" to hide all auto-routed and manually routed cables.**
 All cable connections on devices will be indicated with colored dots in the jacks, and no cables will be displayed.

See "Hiding cables" for more details.

Mouse Knob Range

This lets you adjust the response sensitivity of the various knobs in Reason when manipulating them with the mouse. A higher sensitivity gives a higher degree of precision. You can choose between Normal, Precise and Very Precise.

Theme

Here you can choose from a couple of different visual themes, i.e. how the user interface should be visually presented. The selected theme/color affects the Browser areas. See "About different Themes".

! **Note that you have to restart Reason Rack Plugin for the new Theme to take effect.**

Reason Rack Plugin info

Online Help

Clicking this button opens up the on-line Help system in your default web browser. The Reason Rack Plugin Help contains detailed information about all functions in Reason Rack Plugin. You can choose to browse for information, either from the Table of Contents (TOC), Index or Search tabs in the Help system.

Download Manuals

Clicking this takes you to the support section on Reason Studios, where you can download manuals in pdf format.

About

Clicking this opens up a dialog that informs you about the Reason Rack Plugin version and the people behind it.

Rack Extensions and Content

In this section you can manage your Rack Extensions and your User Account.

User Account

Clicking this button launches your default web browser and brings you to your products page on the Reason Studios web site. Here you can view your list of owned products, including Rack Extensions and download ReFills.

Open Companion

Clicking this button opens the Reason+ Companion application, where you can install Rack Extensions, download Sound Packs etc.

Open Companion on Startup

Check this box to automatically open the Reason Companion application when you launch Reason Rack Plugin.

Go to Shop

Clicking this button starts up your default web browser and takes you to the Reason Studios Shop, where you can get more Rack Extension devices for your Reason Rack Plugin rack and also purchase additional ReFills.

(Additional Content)

Here is information regarding the status of the additional (optional) content that you could download and install in Reason and Reason Rack Plugin. If any additional content is missing, this will be displayed here and you can open the stand-alone version of Reason and download it form the Manage Content item on the Window menu.

Audio

Render audio using host buffer size setting

The “Render audio using host buffer size setting” function should be selected (checked) for best plugin performance. When selected, the audio batches are rendered internally in Reason Rack Plugin according to the set buffer size of the host.



For example, if you are using a buffer size of 512 samples, each audio batch will be 512 samples also in Reason Rack Plug. Raising the buffer size will let Reason Rack Plugin process larger audio batches in one go, which is often more efficient.

If unchecked (off), all audio batches are rendered internally in Reason Rack Plugin at a fixed buffer size of 64 samples - regardless of the audio buffer size setting in your host DAW. This might be desirable if you are using feedback signal routings and CV connections in your songs, and want the internal latency of those connections to be fixed at a short value all the time. This might result in performance problems, though.

Account and Authorization

Here you can see the current status of your Reason Rack Plugin authorization. For example, here is displayed if you are running Reason Rack Plugin with long-term authorization, and which Reason Studios account you are using.

Please refer to "About the Authorization system" in the Introduction chapter for more details regarding authorization.

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